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Applications of Signal Processing to Audio and Acoustics, 1997. 1997 IEEE ASSP Workshop on , 19-1997

Page(s): 4 pp.

[\[Abstract\]](#) [\[PDF Full-Text \(460 KB\)\]](#) **IEEE CNF****2 Feedback cancellation in hearing aids: results from a computer simulation***Kates, J.M.;*

Signal Processing, IEEE Transactions on [see also Acoustics, Speech, and Signal Processing, IEEE Transactions on] , Volume: 39 Issue: 3 , March 1991

Page(s): 553 -562

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Speech and Audio Processing, IEEE Transactions on , Volume: 8 Issue: 4 , July 2000

Page(s): 443 -453

[\[Abstract\]](#) [\[PDF Full-Text \(344 KB\)\]](#) **IEEE JNL****4 Steady-state analysis of continuous adaptation systems for hearing aids with a delay cancellation path***Siqueira, M.G.; Alwan, A.A.;*

Signals, Systems &amp; Computers, 1998. Conference Record of the Thirty-Second Asilomar Conference Volume: 1 , 1-4 Nov. 1998

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**5 Feedback cancellation in hearing aids using constrained adaptation***Kates, J.M.;*

Applications of Signal Processing to Audio and Acoustics, 1999 IEEE Workshop on , 17-20 Oct. 1999

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Speech and Audio Processing, IEEE Transactions on , Volume: 9 Issue: 8 , Nov. 2001

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[\[Abstract\]](#) [\[PDF Full-Text \(178 KB\)\]](#) [IEEE JNL](#)**8 Feedback cancellation in hearing aids***Kates, J.M.;*

Acoustics, Speech, and Signal Processing, 1990. ICASSP-90., 1990 International Conference on , 3-1990

Page(s): 1125 -1128 vol.2

[\[Abstract\]](#) [\[PDF Full-Text \(280 KB\)\]](#) [IEEE CNF](#)**9 Measurement and adaptive suppression of acoustic feedback in hearing aids***Bustamante, D.K.; Worrall, T.L.; Williamson, M.J.;*

Acoustics, Speech, and Signal Processing, 1989. ICASSP-89., 1989 International Conference on , 23-1989

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**12 Subband adaptive filtering applied to acoustic feedback reduction in hearing aids**  
*Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S.; Gao, S.;*  
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**13 Some notes on feedback suppression with adaptive filters in hearing aids**  
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**14 An efficient feedback cancellation for multiband compression hearing aids**  
*Young-Cheol Park; Dong-Wook Kim; In-Young Kim;*  
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**15 Subband signal processing for hearing aids**  
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05407070 E.I. No: EIP99104865423

**Title: Novel approach of adaptive feedback cancellation for hearing aids**

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.

Corporate Source: Univ of California, Los Angeles, CA, USA

Conference Title: Proceedings of the 1999 IEEE International Symposium on Circuits and Systems, ISCAS '99

Conference Location: Orlando, FL, USA Conference Date: 19990530-19990602

E.I. Conference No.: 55489

Source: Proceedings - IEEE International Symposium on Circuits and Systems v 3 1999. p III-195 - III-198

Publication Year: 1999

CODEN: PICSDI ISSN: 0271-4310 ISBN: 0-7803-5471-0

Language: English

**Title: Novel approach of adaptive feedback cancellation for hearing aids**

Abstract: In this paper, a **band - limited adaptive adaptive feedback cancellation** algorithm for **hearing aids** is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

Descriptors: Adaptive **filtering** ; Adaptive algorithms; Oscillations; Hearing aids; Transfer functions; Speech intelligibility; Acoustic properties

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05222884 E.I. No: EIP99024558325

**Title: Compact, ultra low power, programmable continuous-time filter banks for feedback cancellation in hearing aid**

Author: Nair, Kavita; Harjani, Ramesh

Corporate Source: Univ of Minnesota, Minneapolis, MN, USA

Conference Title: Proceedings of the 1999 12th International Conference on VLSI Design

Conference Location: Goa, India Conference Date: 19990107-19990110

E.I. Conference No.: 49707

Source: Proceedings of the IEEE International Conference on VLSI Design 1999. IEEE Comp Soc, Los Alamitos, CA, USA. p 55-60

Publication Year: 1999

CODEN: PIVDEZ

Language: English

**Title: Compact, ultra low power, programmable continuous-time filter banks for feedback cancellation in hearing aid**

Abstract: This paper describes the design of a compact, ultra low power continuous time programmable **filter** . Compact programmable **filters** are required for a number of applications. A particular application is in **feedback cancellation filters in hearing aids** . Here, a **feedback cancellation filter** that matches the open loop transfer function is used to suppress acoustic oscillations. Because of the complexity of the transfer function the number of poles in the cancellation **filter** is fairly large. To realize an integrated cancellation **filter** an ultra-low power transconductance cell with large linear range has been designed. Both measurement...

...implementation and show that there is a significant savings in area as

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well. The complete **filter** has been designed, implemented and fabricated in an 2 mu CMOS technology. (Author abstract) 17...

Descriptors: Digital **filters** ; Low pass **filters** ; Hearing aids; Feedback control; Transfer functions; Electric network synthesis; Transconductance; Capacitors; CMOS integrated circuits; Computer...

Identifiers: Programmable-time **filter** banks; Feedback cancellation **filters**

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11280314 Genuine Article#: 628PN No. References: 29

Title: Band - limited **feedback cancellation with a modified filtered -X LMS algorithm for hearing aids**

Author(s): Chi HF (REPRINT) ; Gao SX; Soli SD; Alwan A

Corporate Source: Virata Corp, 2700 San Tomas Expressway/Santa

Clara//CA/95051 (REPRINT); Virata Corp, Santa Clara//CA/95051; House Ear

Res Inst, Los Angeles//CA/90057; Univ Calif Los Angeles, Dept Elect

Engn, Los Angeles//CA/90095

Journal: SPEECH COMMUNICATION, 2003, V39, N1-2 (JAN), P147-161

ISSN: 0167-6393 Publication date: 20030100

Publisher: ELSEVIER SCIENCE BV, PO BOX 211, 1000 AE AMSTERDAM, NETHERLANDS

Language: English Document Type: ARTICLE (ABSTRACT AVAILABLE)

Title: Band - limited **feedback cancellation with a modified filtered -X LMS algorithm for hearing aids**

...Abstract: not provide satisfactory performance for reducing feedback oscillation in hearing aids. In this paper, a **band - limited** adaptive feedback cancellation algorithm using normalized **filtered -X LMS** techniques is proposed that provides good cancellation efficiency, convergence behavior and better output...

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06238725 E.I. No: EIP02517277836

**Title:** Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids  
**Author:** Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer  
**Corporate Source:** Virata Corporation, Santa Clara, CA 95051, United States  
**Source:** Speech Communication v 39 n 1-2 January 2003. p 147-161  
**Publication Year:** 2003  
**CODEN:** SCOMDH **ISSN:** 0167-6393  
**Language:** English

**Title:** Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids  
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**Descriptors:** Hearing aids ; Feedback; Oscillations; Acoustic waves; Adaptive filtering; Algorithms; Computer simulation  
**Identifiers:** Band - limited feedback cancellation

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**TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)**

**Author:** TUNCER, T. ENGIN

**Degree:** PH.D.

**Year:** 1993

**Corporate Source/Institution:** BOSTON UNIVERSITY (0017)

**Source:** VOLUME 54/02-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 1023. 187 PAGES

**TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)**

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Echo disturbances in speaker /microphone systems is a major problem. The problems of subband echo cancellers are discussed and...

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7549406 INSPEC Abstract Number: A2003-08-8770J-005, B2003-04-7520E-019

**Title:** Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids

**Author(s):** Hsiang-Feng Chi; Gao, S.X.; Soli, S.D.; Alwan, A.

**Author Affiliation:** Virata Corp., Santa Clara, CA, USA

June 27, 2003

Journal: Speech Communication vol.39, no.1-2 p.147-61  
Publisher: Elsevier,  
Publication Date: Jan. 2003 Country of Publication: Netherlands  
CODEN: SCOMDH ISSN: 0167-6393  
SICI: 0167-6393(200301)39:1/2L:147:BLFC;1-X  
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03840589 JICST ACCESSION NUMBER: 99A0045892 FILE SEGMENT: JICST-E  
**Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .**  
UCHIYAMA MICHIAKI (1); TOYAMA MIKIO (1); HIRAI TOORU (2)  
(1) Kogakuin Univ.; (2) Yamaha Corp.  
Denshi Joho Tsushin Gakkai Gijutsu Kenkyu Hokoku(IEIC Technical Report (Institute of Electronics, Information and Communication Engineers), 1998, VOL.98,NO.277(EA98 59-64), PAGE.41-46, FIG.10, REF.2  
JOURNAL NUMBER: S0532BBG  
UNIVERSAL DECIMAL CLASSIFICATION: 621.391.8 621.372.54  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
DOCUMENT TYPE: Journal  
ARTICLE TYPE: Original paper  
MEDIA TYPE: Printed Publication

**Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .**  
ABSTRACT: As acoustic collaboration technologies are being developed, speaker -phone control, such as howling control in a closed loop has been important. In this...  
BROADER DESCRIPTORS: band stop filter...

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**Control of Antiresonance by a Variable Damping Active Resonator.**  
MATSUHIRA HIROSHI (1); SATO SUSUMU (1); TSUJIMOTO IKUO (2)  
(1) Kyoto Univ., Faculty of Engineering; (2) Kyoto Univ., Graduate School  
Nippon Kikai Gakkai Ronbunshu. C(Transactions of the Japan Society of Mechanical Engineers. C), 1993, VOL.59,NO.562, PAGE.1824-1829, FIG.12, REF.5  
JOURNAL NUMBER: F0045BAL ISSN NO: 0387-5024  
UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
DOCUMENT TYPE: Journal

June 27, 2003

ARTICLE TYPE: Original paper  
MEDIA TYPE: Printed Publication

...ABSTRACT: controlled by feedback of sound pressure detected by a microphone in the resonator to a **speaker** on the wall of the resonator cavity. When the feedback gain is increased, the damping...

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**Title:** Band - limited feedback cancellation with a modified filtered -X LMS algorithm for hearing aids  
**Author:** Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.; Alwan, Abeer  
**Corporate Source:** Virata Corporation, Santa Clara, CA 95051, United States  
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**Publication Year:** 2003  
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**Identifiers:** Band - limited feedback cancellation

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**TWO-BAND QMF BANKS, BUTTERWORTH WAVELETS, TIME-VARYING FILTER BANKS AND ADAPTIVE ECHO CANCELLATION (QMF BANKS)**

**Author:** TUNCER, T. ENGIN  
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**Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .**  
UCHIYAMA MICHIAKI (1); TOYAMA MIKIO (1); HIRAI TOORU (2)  
(1) Kogakuin Univ.; (2) Yamaha Corp.  
Denshi Joho Tsushin Gakkai Gijutsu Kenkyu Hokoku(IEIC Technical Report (Institute of Electronics, Information and Communication Engineers), 1998, VOL.98,NO.277(EA98 59-64), PAGE.41-46, FIG.10, REF.2  
JOURNAL NUMBER: S0532BBG  
UNIVERSAL DECIMAL CLASSIFICATION: 621.391.8 621.372.54  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
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**Howling Suppression for a Tele-Communication System using both a Time-Variant Notch Filter and an Acoustic Echo Canceller .**  
ABSTRACT: As acoustic collaboration technologies are being developed, **speaker** -phone control, such as howling control in a closed loop has been important. In this...  
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01786012 JICST ACCESSION NUMBER: 93A0619340 FILE SEGMENT: JICST-E  
**Control of Antiresonance by a Variable Damping Active Resonator.**  
MATSUHIRA HIROSHI (1); SATO SUSUMU (1); TSUJIMOTO IKUO (2)  
(1) Kyoto Univ., Faculty of Engineering; (2) Kyoto Univ., Graduate School  
Nippon Kikai Gakkai Ronbunshu. C(Transactions of the Japan Society of Mechanical Engineers. C), 1993, VOL.59,NO.562, PAGE.1824-1829, FIG.12, REF.5  
JOURNAL NUMBER: F0045BAL ISSN NO: 0387-5024  
UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
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June 27, 2003

MEDIA TYPE: Printed Publication

...ABSTRACT: controlled by feedback of sound pressure detected by a microphone in the resonator to a **speaker** on the wall of the resonator cavity. When the feedback gain is increased, the damping...

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**Title: Blind separation and blind deconvolution: an information-theoretic approach**

Author: Bell, Anthony J.; Sejnowski, Terrence J.  
Corporate Source: Salk Inst, La Jolla, CA, USA  
Conference Title: Proceedings of the 1995 International Conference on Acoustics, Speech, and Signal Processing. Part 5 (of 5)  
Conference Location: Detroit, MI, USA Conference Date: 19950509-19950512  
E.I. Conference No.: 43559  
Source: Special Sessions ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing - Proceedings v 5 1995. IEEE, Piscataway, NJ, USA, 95CH35732. p 3415-3418  
Publication Year: 1995  
CODEN: IPRODJ ISSN: 0736-7791  
Language: English

...Abstract: By using a new algorithm, nearly perfect separation of up to 10 digitally mixed human **speakers** is achieved. This performance is significantly better than any previous algorithm. When used for deconvolution, the technique automatically **cancels echoes** and reverberations and reverses the effects of **low - pass filtering** . 9  
Refs.

Descriptors: Learning algorithms; Digital signal processing; Matrix algebra; **Low pass filters** ; Codes (symbols); Statistics; Calculations; Speech processing; Echo suppression; Reverberation

14/3,K/2 (Item 2 from file: 8)  
DIALOG(R)File 8:EI Compendex(R)  
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03848202 E.I. No: EIP94041267278

**Title: Adaptive active noise control system with howling canceler (2-microphone transfer function ratio system)**

Author: Suzuki, Seiichirou; Hayashi, Takurou  
Source: Nippon Kikai Gakkai Ronbunshu, C Hen/Transactions of the Japan Society of Mechanical Engineers, Part C v 60 n 569 Jan 1994. p 169-174  
Publication Year: 1994  
CODEN: NKCHDB ISSN: 0387-5024  
Language: Japanese

...Abstract: a howling canceler system for active noise control in a duct. The effect of howling **canceler** employing an **echo canceler** or two kinds of 2-microphone systems (one is a 2-Microphone delay system and ...

...this howling canceler, particularly utilizing the 2-Microphone transfer function ratio system, is examined. A **filtered -X** algorithm was used, because an acoustical transfer function from the secondary source to the...

...the discrete system in the low frequency range because of the low efficiency of the **speaker** in this range. In this case, the **high - pass filter** effect of the 2-Microphone howling canceler system was useful for achieving stable control. This...

...Identifiers: active noise control system; Howling canceler system; 2-microphone transfer function ratio system; Noise reduction; **Filtered x**-algorithm; **Speaker** ; Fan noise; **High pass filter**

14/3,K/3 (Item 1 from file: 35)

June 27, 2003

DIALOG(R)File 35:Dissertation Abs Online  
(c) 2003 ProQuest Info&Learning. All rts. reserv.

908363 ORDER NO: AAD86-02910

**EVALUATION OF PHASE COMPENSATION FOR ENHANCING THE SIGNAL PROCESSING  
CAPABILITIES OF HEARING AIDS IN SITU**

Author: PREVES, DAVID ALLAN

Degree: PH.D.

Year: 1985

Corporate Source/Institution: UNIVERSITY OF MINNESOTA (0130)

Source: VOLUME 46/12-B OF DISSERTATION ABSTRACTS INTERNATIONAL.

PAGE 4327. 203 PAGES

**EVALUATION OF PHASE COMPENSATION FOR ENHANCING THE SIGNAL PROCESSING  
CAPABILITIES OF HEARING AIDS IN SITU**

A hearing aid worn in situ was treated as a feedback control loop and was found to meet the Nyquist criteria for stability. At high gain settings, acoustic...

...to recognize. Two behavioral investigations addressed the question of whether adding phase compensation to a hearing aid significantly changed recognition scores using the speech materials selected from the literature review. The first...

...impaired listeners. The second study tested whether high frequency hearing loss could be simulated by low pass filtering the taped playback of speech stimuli and whether adding phase compensation to a hearing aid resulted in confusion patterns more closely resembling that of normal hearing listeners than that produced...

14/3,K/4 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

4675181 INSPEC Abstract Number: A9413-4370-004, B9407-6130-006,  
C9407-1250C-003

Title: A real-time isolated word recognizer for telephone input

Author(s): Yato, F.; Kuroiwa, S.; Takeda, K.; Yamamoto, S.; Owa, K.; Shozakai, M.

Author Affiliation: KDD Kamifukuoka R&D Labs., Saitama, Japan

Journal: Journal of the Acoustical Society of Japan (E) vol.15, no.2  
p.87-96

Publication Date: March 1994 Country of Publication: Japan

CODEN: JASED2 ISSN: 0388-2861

Language: English

Subfile: A B C

...Abstract: telephone network, we developed feature extraction and a word detection algorithm. These techniques use wide band - pass filter outputs which are generally employed to decide whether speech is voiced or unvoiced. To achieve a friendly interface, the system can accept user input at any time by using an echo canceller and the new word detection algorithm. Finally, the recogniser is evaluated using a large telephone voice database consisting of more than 500 speakers.

...Identifiers: wide band - pass filter outputs...  
... echo canceller ;

14/3,K/5 (Item 1 from file: 94)

DIALOG(R)File 94:JICST-EPlus

(c)2003 Japan Science and Tech Corp(JST). All rts. reserv.

01877738 JICST ACCESSION NUMBER: 93A0740540 FILE SEGMENT: JICST-E  
A Study on the Characteristics of 2-microphone Transfer Function Ratio

June 27, 2003

**System.**

HAYASHI TAKURO (1); SUZUKI SEIICHIRO (2); (2) Toshiba Corp.  
Nippon Kikai Gakkai Kikai Rikigaku, Keisoku Seigyo Koen Ronbunshu, 1993,  
VOL.1993,NO.B, PAGE.127-132, FIG.12, REF.6  
JOURNAL NUMBER: L1497AAE  
UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
DOCUMENT TYPE: Conference Proceeding  
ARTICLE TYPE: Original paper  
MEDIA TYPE: Printed Publication

...ABSTRACT: howling canceler system for active noise control in a duct.  
The effect of the howling **canceler** employing an **echo canceler** or  
two kinds of 2-microphone systems (one is 2-microphone delay system and  
the...

...using this howling canceler, especially using 2-microphone transfer  
function ratio system, is examined. A **Filtered -X** adaptive **filter**  
algorithm was used because acoustical transfer function from the  
secondary source to the evaluation microphone...

...the discrete system in the low frequency range because of the low  
efficiency of the **speaker** in this range. In this case, the **high**  
**pass filter** effect of the 2-microphone howling canceler system is  
useful for achieving stable control. The...

...delay through the control system was analyzed. This time delay concerns  
the causality of FIR **filter**. Some distance between detecting  
microphone and secondary source to keep this causality was confirmed  
from...

...DESCRIPTORS: FIR **filter** ;  
...BROADER DESCRIPTORS: digital **filter** ; ...

... **filter** (signal...

... **filter** ;

14/3,K/6 (Item 2 from file: 94)  
DIALOG(R)File 94:JICST-EPlus  
(c)2003 Japan Science and Tech Corp(JST). All rts. reserv.

01732128 JICST ACCESSION NUMBER: 93A0427291 FILE SEGMENT: JICST-E  
**A Study on Adaptive Active Noise Control System with Howling Canceller.**  
SUZUKI SEIICHIRO (1); HAYASHI TAKURO (1)  
(1) Toshiba Corp.  
Nippon Kikai Gakkai Tsujo Sokai Koenkai Koen Ronbunshu(Proceedings of the  
International Sessions JSME Spring Annual Meeting), 1993,  
VOL.70th,NO.Pt 3, PAGE.146-148, FIG.6, REF.4  
JOURNAL NUMBER: X0588AAU  
UNIVERSAL DECIMAL CLASSIFICATION: 534.2+534.8  
LANGUAGE: Japanese COUNTRY OF PUBLICATION: Japan  
DOCUMENT TYPE: Conference Proceeding  
ARTICLE TYPE: Short Communication  
MEDIA TYPE: Printed Publication

...ABSTRACT: howling canceler system for active noise control in a duct.  
The effect of the howling **canceler** employing an **echo canceler** or  
2-microphone system was confirmed from the viewpoint of eliminating  
sound pressure signal from...

...noise reduction effect of the adaptive control system using this howling  
canceler is examined. A **Filtered -X** algorithm was used because an  
acoustical transfer function from the secondary source to the...

June 27, 2003

...the discrete system in the low frequency range because of the low efficiency of the **speaker** in this range. In this case, the **high pass filter** effect of the 2-microphone howling canceler system is useful for achieving stable control. This...

June 27, 2003

**24/3,K/1 (Item 1 from file: 8)**  
DIALOG(R)File 8:Ei Compendex(R)  
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

06238725 E.I. No: EIP02517277836

**Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids**

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D. ; Alwan, Abeer  
Corporate Source: Virata Corporation, Santa Clara, CA 95051, United States

Source: Speech Communication v 39 n 1-2 January 2003. p 147-161

Publication Year: 2003

CODEN: SCOMDH ISSN: 0167-6393

Language: English

**Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids**

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D. ; Alwan, Abeer

...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in **hearing aids**. In this paper, a band-limited adaptive feedback cancellation algorithm using normalized filtered-X LMS...

Descriptors: **Hearing aids** ; Feedback; Oscillations; **Acoustic waves**; Adaptive filtering; Algorithms; Computer simulation

**24/3,K/2 (Item 2 from file: 8)**  
DIALOG(R)File 8:Ei Compendex(R)  
(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

05407070 E.I. No: EIP99104865423

**Title: Novel approach of adaptive feedback cancellation for hearing aids**

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.

Corporate Source: Univ of California, Los Angeles, CA, USA

Conference Title: Proceedings of the 1999 IEEE International Symposium on Circuits and Systems, ISCAS '99

Conference Location: Orlando, FL, USA Conference Date: 19990530-19990602

E.I. Conference No.: 55489

Source: Proceedings - IEEE International Symposium on Circuits and Systems v 3 1999. p III-195 - III-198

Publication Year: 1999

CODEN: PICSDI ISSN: 0271-4310 ISBN: 0-7803-5471-0

Language: English

**Title: Novel approach of adaptive feedback cancellation for hearing aids**

Author: Chi, Hsiang-Feng; Gao, Shawn X.; Soli, Sigfrid D.

Abstract: In this paper, a band-limited adaptive adaptive feedback cancellation algorithm for **hearing aids** is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

Descriptors: Adaptive filtering; Adaptive algorithms; Oscillations; **Hearing aids** ; Transfer functions; Speech intelligibility; Acoustic properties

**24/3,K/3 (Item 3 from file: 8)**  
DIALOG(R)File 8:Ei Compendex(R)  
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04729562 E.I. No: EIP97063700503

**Title: Subband adaptive filtering applied to acoustic feedback reduction**

June 27, 2003

in hearing aids

Author: Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S. ; Gao, S.

Corporate Source: Univ of California, Los Angeles, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 30th Asilomar Conference on Signals, Systems & Computers. Part 1 (of 2)

Conference Location: Pacific Grove, CA, USA Conference Date: 19961103-19961106

E.I. Conference No.: 46513

Source: Conference Record of the Asilomar Conference on Signals, Systems & Computers v 1 1997. IEEE, Los Alamitos, CA, USA, 96CB36004. p 788-792

Publication Year: 1997

CODEN: CCSCE2 ISSN: 1058-6393

Language: English

**Title: Subband adaptive filtering applied to acoustic feedback reduction in hearing aids**

Author: Siqueira, M.G.; Speece, R.; Petsalis, E.; Alwan, A.; Soli, S. ; Gao, S.

Abstract: Acoustic feedback is a problem in hearing aids that contain a substantial amount of gain, hearing aids that are used in conjunction with vented or open molds, and in-the-ear hearing aids. Acoustic feedback is both annoying and reduces the maximum usable gain of hearing - aid devices. This paper models the time-varying acoustic feedback path for hearing aids before performing a systematic evaluation of acoustic feedback reduction techniques through perceptual experiments. It is...

Descriptors: Adaptive filtering; Feedback; Hearing aids ; Acoustic signal processing; Algorithms; Computational complexity

24/3,K/4 (Item 4 from file: 8)

DIALOG(R)File 8: Ei Compendex(R)

(c) 2003 Elsevier Eng. Info. Inc. All rts. reserv.

04560676 E.I. No: EIP96110412592

**Title: Calibration, optimization, and DSP implementation of microphone array for speech processing**

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

Corporate Source: UCLA, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 9th IEEE Workshop on VLSI Signal Processing

Conference Location: San Francisco, CA, USA Conference Date: 19961030-19961101

E.I. Conference No.: 45510

Source: IEEE Workshop on VLSI Signal Processing, Proceedings 1996. IEEE, Piscataway, NJ, USA. p 221-228

Publication Year: 1996

CODEN: 85PYA8

Language: English

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

Abstract: For various audio, teleconference, hearing aid, and voice recognition applications, a microphone array is known to be an effective method to...

24/3,K/5 (Item 5 from file: 8)

DIALOG(R)File 8: Ei Compendex(R)

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04533861 E.I. No: EIP96103369592

**Title: High performance microphone array system for hearing aid applications**

June 27, 2003

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.F.; Gao, S.

Corporate Source: UCLA Los Angeles, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 IEEE International Conference on Acoustics, Speech, and Signal Processing, ICASSP. Part 6 (of 6)

Conference Location: Atlanta, GA, USA Conference Date: 19960507-19960510

E.I. Conference No.: 45447

Source: ICASSP, IEEE International Conference on Acoustics, Speech and Signal Processing - Proceedings v 6 1996. IEEE, Piscataway, NJ, USA, 96CB35903. p 3197-3200

Publication Year: 1996

CODEN: IPRODJ ISSN: 0736-7791

Language: English

**Title: High performance microphone array system for hearing aid applications**

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.F.; Gao, S.

Abstract: Microphone array technology has been proposed for various audio, teleconference, and hearing aid applications. By forming a focused beam toward the desired speech source, attenuating background noises and...

...In this paper, we present a high performance prototype PC-based microphone array system for hearing aid applications. Algorithms for maximum energy criterion array weight design needed in the speech processing mode...

Descriptors: Acoustic arrays; Hearing aids; Microphones; Algorithms; Signal interference; Interference suppression; Matrix algebra; Signal filtering and prediction; Acoustic signal processing...

24/3,K/6 (Item 6 from file: 8)

DIALOG(R) File 8: Ei Compendex(R)

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04528235 E.I. No: EIP96103361906

**Title: Microphone array for hearing aid and speech enhancement applications**

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.; Gao, S.

Corporate Source: UCLA, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 International Conference on Application-Specific Systems, Architectures and Processors

Conference Location: Chicago, IL, USA Conference Date: 19960819-19960821

E.I. Conference No.: 45391

Source: International Conference on Application-Specific Systems, Architectures and Processors, Proceedings 1996. IEEE, Piscataway, NJ, USA. p 231-239

Publication Year: 1996

CODEN: 002451

Language: English

**Title: Microphone array for hearing aid and speech enhancement applications**

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.; Gao, S.

Abstract: Microphone array technology has been proposed for various audio, teleconference, hearing aid, and voice recognition applications. By forming a focused beam toward the desired speech source, attenuating...

...reverberations and competing interferences. We present a prototype

June 27, 2003

PC-based microphone array system designed for **hearing aid** applications but also applicable to other tasks. In section 1, a multiple channel microphone array...

**24/3,K/7 (Item 7 from file: 8)**

DIALOG(R)File 8:Ei Compendex(R)

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04507867 E.I. No: EIP96093340787

**Title: Novel DSP system for microphone array applications**

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

Corporate Source: UCLA, Los Angeles, CA, USA

Conference Title: Proceedings of the 1996 IEEE International Symposium on Circuits and Systems, ISCAS. Part 2 (of 4)

Conference Location: Atlanta, GA, USA Conference Date: 19960512-19960515

E.I. Conference No.: 45321

Source: Circuits and Systems Connecting the World Proceedings - IEEE International Symposium on Circuits and Systems v 2 1996. IEEE, Piscataway, NJ, USA, 96CB35876. p 201-204

Publication Year: 1996

CODEN: PICSDI ISSN: 0271-4310

Language: English

Author: Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.; Soli, S.D. ; Gao, S.

...Abstract: Among these, digital microphone array has been proposed for sonar, audio, teleconferencing, multi-media, and **hearing aid** applications left bracket 1, 2, 4, 3, 5 right bracket . A microphone array can enhance...

...sources in various applications. We present a prototype PC-based microphone array system designed for **hearing aid** applications but also applicable to other tasks. In section 1, a multiple channel microphone array...

**24/3,K/8 (Item 8 from file: 8)**

DIALOG(R)File 8:Ei Compendex(R)

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04368207 E.I. No: EIP96033107316

**Title: Development of a prototype portable binaural digital hearing aid**

Author: Soli, Sigfrid D.

Corporate Source: House Ear Inst, Los Angeles, CA, USA

Conference Title: Proceedings of the 1995 3rd URSI International Symposium on Signals, Systems and Electronics, ISSSE'95

Conference Location: San Francisco, CA, USA Conference Date: 19951025-19951027

E.I. Conference No.: 44409

Source: Conference Proceedings of the International Symposium on Signals, Systems and Electronics 1995. IEEE, Piscataway, NJ, USA, 95TH8047. p 381

Publication Year: 1995

CODEN: 002316

Language: English

**Title: Development of a prototype portable binaural digital hearing aid**

Author: Soli, Sigfrid D.

Abstract: In order for a **hearing aid** user to perform binaural directional hearing in noisy environments, it is important to maintain at audible levels the binaural cues that would be present if the **hearing**

June 27, 2003

aid were not in place. In view of this, a wearable prototype digital hearing aid and personal computer (PC) based methods of digital design for evaluation of the effectiveness of...

...fitting algorithms, methods and accuracy of fittings, and the hardware and software comprising the binaural hearing aid are discussed, as well as the results from field trials with the portable processors.

Descriptors: Hearing aids ; Personal computers; Digital filters; Electric network synthesis; Program processors; Microprocessor chips; Signal filtering and prediction...

Identifiers: Prototype portable binaural digital hearing aid ; Digital filter design; Body worn prototype processor; Digital signal processing chip; Binaural fitting; Hearing aid equalization; Hearing loss compensation

24/3,K/9 (Item 1 from file: 65)

DIALOG(R)File 65:Inside Conferences

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01486120 INSIDE CONFERENCE ITEM ID: CN014751276

Digital Signal Processing for An Ear Leel Binaural Hearing Aid  
Soli, S.

CONFERENCE: Signals, systems and electronics-International symposium; 3rd ISSE -INTERNATIONAL SYMPOSIUM-, 1995; 3rd P: 381-381

IEEE, 1995

ISBN: 0780325168

LANGUAGE: English DOCUMENT TYPE: Conference Ppaers

CONFERENCE SPONSOR: Union Radio-Scientifique Internationale

CONFERENCE LOCATION: San Francisco, CA

CONFERENCE DATE: Oct 1995 (19951) (19951)

Digital Signal Processing for An Ear Leel Binaural Hearing Aid  
Soli, S.

24/3,K/10 (Item 1 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7549406 INSPEC Abstract Number: A2003-08-8770J-005, B2003-04-7520E-019

Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D. ; Alwan, A.

Author Affiliation: Virata Corp., Santa Clara, CA, USA

Journal: Speech Communication vol.39, no.1-2 p.147-61

Publisher: Elsevier,

Publication Date: Jan. 2003 Country of Publication: Netherlands

CODEN: SCOMDH ISSN: 0167-6393

SICI: 0167-6393(200301)39:1/2L.147:BLFC;1-X

Material Identity Number: C760-2002-007

U.S. Copyright Clearance Center Code: 0167-6393/03/\$30.00

Language: English

Subfile: A B

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Title: Band-limited feedback cancellation with a modified filtered-X LMS algorithm for hearing aids

Author(s): Hsiang-Feng Chi; Gao, S.X.; Soli, S.D. ; Alwan, A.

...Abstract: wideband adaptive feedback cancellation techniques do not provide satisfactory performance for reducing feedback oscillation in hearing aids . A band-limited adaptive feedback cancellation algorithm using normalized filtered-X LMS techniques is proposed...

...Descriptors: hearing aids ;

...Identifiers: hearing aids ;

June 27, 2003

24/3,K/11 (Item 2 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

7180527 INSPEC Abstract Number: A2002-06-8736-002, B2002-03-7520E-020

**Title: On application of adaptive decorrelation filtering to assistive listening**

Author(s): Yunxin Zhao; Kuan-Chieh Yen; **Soli, S.** ; Shawn Gao; Vermiglio, A.

Author Affiliation: Dept. of Comput. Eng. & Comput. Sci., Missouri Univ., Columbia, MO, USA

Journal: Journal of the Acoustical Society of America vol.111, no.2 p.1077-85

Publisher: Acoust. Soc. America through AIP,

Publication Date: Feb. 2002 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

SICI: 0001-4966(200202)111:2L:1077:AADF;1-I

Material Identity Number: J001-2002-002

U.S. Copyright Clearance Center Code: 0001-4966/2002/111(2)/1077/9/\$18.00

Language: English

Subfile: A B

Copyright 2002, IEE

Author(s): Yunxin Zhao; Kuan-Chieh Yen; **Soli, S.** ; Shawn Gao; Vermiglio, A.

...Descriptors: **hearing aids** ;

24/3,K/12 (Item 3 from file: 2)

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

6430420 INSPEC Abstract Number: A2000-02-8770J-015, B2000-01-7520E-032

**Title: A novel approach of adaptive feedback cancellation for hearing aids**

Author(s): Hsiang-Feng Chi; Gao, S.X.; **Soli, S.D.**

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles, CA, USA

Conference Title: ISCAS'99. Proceedings of the 1999 IEEE International Symposium on Circuits and Systems VLSI (Cat. No.99CH36349) Part vol.3 p.195-8 vol.3

Publisher: IEEE, Piscataway, NJ, USA

Publication Date: 1999 Country of Publication: USA 6 vol. (liv+565+717+568+604+647+527) pp.

ISBN: 0 7803 5471 0 Material Identity Number: XX-1999-01882

U.S. Copyright Clearance Center Code: 0 7803 5471 0/99/\$10.00

Conference Title: ISCAS'99. Proceedings of the 1999 IEEE International Symposium on Circuits and Systems. VLSI

Conference Date: 30 May-2 June 1999 Conference Location: Orlando, FL, USA

Language: English

Subfile: A B

Copyright 1999, IEE

**Title: A novel approach of adaptive feedback cancellation for hearing aids**

Author(s): Hsiang-Feng Chi; Gao, S.X.; **Soli, S.D.**

Abstract: In this paper, a band-limited adaptive feedback cancellation algorithm for **hearing aids** is proposed. Utilizing the characteristics of the feedback oscillation, the algorithm provides better cancellation efficiency...

...Descriptors: **hearing aids**

...Identifiers: **hearing aids** ;

June 27, 2003

**24/3,K/13 (Item 4 from file: 2)**

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5480227 INSPEC Abstract Number: A9705-4360-001, B9703-7810C-003,  
C9703-5585-001

**Title: Modern microphone array for hearing aid and speech processing**

Author(s): Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;  
**Soli, S.D.** ; Gao, S.

Author Affiliation: Dept. of Electr. Eng., California Univ., Los Angeles,  
CA, USA

Journal: Proceedings of the SPIE - The International Society for Optical  
Engineering Conference Title: Proc. SPIE - Int. Soc. Opt. Eng. (USA)  
vol.2846 p.112-21

Publisher: SPIE-Int. Soc. Opt. Eng,

Publication Date: 1996 Country of Publication: USA

CODEN: PSISDG ISSN: 0277-786X

SICI: 0277-786X(1996)2846L:112:MMAH;1-A

Material Identity Number: C574-96279

U.S. Copyright Clearance Center Code: 0 8194 2234 7/96/\$6.00

Conference Title: Advanced Signal Processing Algorithms, Architectures,  
and Implementations VI

Conference Sponsor: SPIE

Conference Date: 6-8 Aug. 1996 Conference Location: Denver, CO, USA

Language: English

Subfile: A B C

Copyright 1997, IEE

**Title: Modern microphone array for hearing aid and speech processing**

Author(s): Wang, A.; Yao, K.; Hudson, R.E.; Korompis, D.; Lorenzelli, F.;  
**Soli, S.D.** ; Gao, S.

Abstract: For various **audio** , teleconference, **hearing aid** , and voice  
recognition applications, a microphone array is known to be an effective  
method to...

...Descriptors: **hearing aids** ;

...Identifiers: **hearing aid applications**

**24/3,K/14 (Item 5 from file: 2)**

DIALOG(R)File 2:INSPEC

(c) 2003 Institution of Electrical Engineers. All rts. reserv.

5103032 INSPEC Abstract Number: A9524-8734-012

**Title: Electrode ranking of "place pitch" and speech recognition in  
electrical hearing**

Author(s): Nelson, D.A.; Van Tasell, D.J.; Schroder, A.C.; **Soli, S.** ;  
Levine, S.

Author Affiliation: Clinical Psychoacoust. Lab., Minnesota Univ.,  
Minneapolis, MN, USA

Journal: Journal of the Acoustical Society of America vol.98, no.4  
p.1987-99

Publication Date: Oct. 1995 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

U.S. Copyright Clearance Center Code: 0001-4966/95/98(4)/1987/13/\$6.00

Language: English

Subfile: A

Copyright 1995, IEE

Author(s): Nelson, D.A.; Van Tasell, D.J.; Schroder, A.C.; **Soli, S.** ;  
Levine, S.

Descriptors: **hearing aids** ;

**24/3,K/15 (Item 6 from file: 2)**

June 27, 2003

DIALOG(R)File 2:INSPEC

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03804369 INSPEC Abstract Number: A91025100

**Title: Acoustic cues for consonant identification by patients who use the Ineraid cochlear implant**

Author(s): Dorman, M.F.; **Soli, S.** ; Dankowski, K.; Smith, L.M.; McCandless, G.; Parkin, J.

Author Affiliation: Arizona State Univ., Tempe, AZ, USA

Journal: Journal of the Acoustical Society of America vol.88, no.5

p.2074-9

Publication Date: Nov. 1990 Country of Publication: USA

CODEN: JASMAN ISSN: 0001-4966

U.S. Copyright Clearance Center Code: 0001-4966/90/112074-06\$00.80

Language: English

Subfile: A

Author(s): Dorman, M.F.; **Soli, S.** ; Dankowski, K.; Smith, L.M.; McCandless, G.; Parkin, J.

Descriptors: **hearing aids** ;

**24/3,K/16 (Item 1 from file: 144)**

DIALOG(R)File 144:Pascal

(c) 2003 INIST/CNRS. All rts. reserv.

15430554 PASCAL No.: 02-0122403

**On application of adaptive decorrelation filtering to assistive listening**

ZHAO Yunxin; YEN Kuan-Chieh; **SOLI Sig** ; GAO Shawn; VERMIGLIO Andy

Department of Computer Engineering and Computer Science, University of Missouri-Columbia, Columbia, Missouri 65211; Beckman Institute and Department of ECE, University of Illinois at Urbana-Champaign, Urbana, Illinois 61801; Human Communication Sciences and Devices Department, House Ear Institute, Los Angeles, California 90057

Journal: The Journal of the Acoustical Society of America, 2002-02, 111 (2) 1077-1085

Language: English

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ZHAO Yunxin; YEN Kuan-Chieh; **SOLI Sig** ; GAO Shawn; VERMIGLIO Andy

English Descriptors: Experimental study; Speech intelligibility; **Hearing aids** ; Decorrelation; **Acoustic** filters; Acoustic signal processing; Adaptive filters

**24/3,K/17 (Item 2 from file: 144)**

DIALOG(R)File 144:Pascal

(c) 2003 INIST/CNRS. All rts. reserv.

15000067 PASCAL No.: 01-0155506

**Method of measuring and preventing unstable feedback in hearing aids**

GAO Shawn X; **SOLI Sigfrid D**

Journal: The Journal of the Acoustical Society of America, 2001-04, 109 (4) p. 1283

Language: English

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**Method of measuring and preventing unstable feedback in hearing aids**

GAO Shawn X; **SOLI Sigfrid D**

English Descriptors: Instrumentation; Measuring methods; **Hearing aids** ; Microphones; **Acoustic** variables measurement

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24/3,K/18 (Item 3 from file: 144)  
DIALOG(R)File 144:Pascal  
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12041469 PASCAL No.: 95-0237256  
**Effects of hearing aids on binaural directional hearing in hearing-impaired individuals**  
GELNETT Donna J; NILSSON Michael J; SOLI Sigfrid D  
House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057  
The 129th Meeting of the Acoustical Society of America (Washington, DC (USA)) 1995-05-30/1995-06-03  
Journal: Journal of the Acoustical Society of America, 1995-05, 97 (5) 3346-3346  
Language: English

Copyright (c) 1995 American Institute of Physics

**Effects of hearing aids on binaural directional hearing in hearing-impaired individuals**  
GELNETT Donna J; NILSSON Michael J; SOLI Sigfrid D  
...spatial separation of the speech and a spectrally matched noise for 25 hearing-impaired binaural hearing aid users. Directional hearing capacity for these individuals often fell within the normal range. Unaided RTSs were elevated 3...  
... that the interaural cues for binaural directional hearing are either inaudible or absent from the hearing aid output. Detailed analyses will be reported with respect to the type of hearing aid, hearing aid transfer function, and degree of hearing loss.

English Descriptors: Experimental study; Hearing ; Hearing aids ; Hearing imp airmen; Directivity; Speech recognition; Noise; Transfer functions; Comparative evaluations

24/3,K/19 (Item 4 from file: 144)  
DIALOG(R)File 144:Pascal  
(c) 2003 INIST/CNRS. All rts. reserv.

12041467 PASCAL No.: 95-0237254  
**Field trials of a portable prototype digital hearing aid**  
GELNETT Donna J; SULLIVAN Jean A; NILSSON Michael J; SOLI Sigfrid D  
House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057  
The 129th Meeting of the Acoustical Society of America (Washington, DC (USA)) 1995-05-30/1995-06-03  
Journal: Journal of the Acoustical Society of America, 1995-05, 97 (5) 3346-3346  
Language: English

Copyright (c) 1995 American Institute of Physics

**Field trials of a portable prototype digital hearing aid**  
GELNETT Donna J; SULLIVAN Jean A; NILSSON Michael J; SOLI Sigfrid D  
...and receivers located in left and right ear modules was built and used in a hearing aid field trial. Eight hearing impaired individuals with moderate to moderately severe hearing losses served as subjects. All subjects had symmetric hearing losses and were experienced binaural hearing aid users. Four binaural hearing aid algorithms were programmed into the processor for evaluation in the field trial. The algorithms all...

English Descriptors: Experimental study; Hearing aids ; Digital systems; Microprocessors; Algorithms; Speech recognition; Noise; Testing

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24/3,K/20 (Item 5 from file: 144)  
DIALOG(R)File 144:Pascal  
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11959943 PASCAL No.: 95-0140407

**Method of signal processing for maintaining directional hearing with hearing aids**

**SOLI Sigfrid D**

Journal: Journal of the Acoustical Society of America, 1995-01, 97 (1)  
733-733

Language: English

Copyright (c) 1995 American Institute of Physics

**Method of signal processing for maintaining directional hearing with hearing aids**

**SOLI Sigfrid D**

24/3,K/21 (Item 6 from file: 144)  
DIALOG(R)File 144:Pascal  
(c) 2003 INIST/CNRS. All rts. reserv.

11756536 PASCAL No.: 94-0627513

**Wideband microphone array for hearing aid preprocessing**

YAO Kung; **SOLI Sigfrid D**; KOROMPIS Dan

Elect. Eng. Dept., Eng. IV, 68-113, UCLA, Los Angeles, CA 91403-1594;  
House Ear Inst., Los Angeles, CA 90057; UCLA, Los Angeles, CA 91403-1594  
The 128th Meeting of the Acoustical Society of America (Austin, Texas  
(USA)) 1994-11-28/1994-12-02

Journal: Journal of the Acoustical Society of America, 1994-11, 96 (5)  
3244-3245

Language: English

Copyright (c) 1994 American Institute of Physics

**Wideband microphone array for hearing aid preprocessing**

YAO Kung; **SOLI Sigfrid D**; KOROMPIS Dan

...hearing and hearing impaired individuals. The feasibility of real-time  
acoustic beamformers with arrays for **hearing aids**, and the advantages  
of this scheme over conventional adaptive schemes will also be discussed.

...English Descriptors: Experimental study; Measuring methods; HEARING  
IMPAIRMENT; Microphones; Signal-to-noise ratio; Constraints; Eigenvalues;  
Numerical solution; **HEARING AIDS**; Speech recognition

24/3,K/22 (Item 7 from file: 144)  
DIALOG(R)File 144:Pascal  
(c) 2003 INIST/CNRS. All rts. reserv.

11484530 PASCAL No.: 94-0322525

**Norms for a headphone simulation of the Hearing in Noise Test: Comparison of physical and simulated spatial separation of sound sources**

NILSSON Michael J; **SOLI Sigfrid D**

House Ear Inst., 2100 West Third St., Los Angeles, CA 90057  
The 127th Meeting of the Acoustical Society of America (Cambridge,  
Massachusetts (USA)) 1994-06-06/1994-06-10

Journal: Journal of the Acoustical Society of America, 1994-05, 95 (5)  
2994-2994

Language: English

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NILSSON Michael J; **SOLI Sigfrid D**

... noise are all lower in the headphone system, attributable to the elimination of room and **speaker** effects. Improvements in SSRTs with spatial separation of the signal and masker were 6.38...

**24/3,K/23 (Item 8 from file: 144)**

DIALOG(R)File 144:Pascal

(c) 2003 INIST/CNRS. All rts. reserv.

11484518 PASCAL No.: 94-0322513

**Method for fitting binaural hearing aids**

GAO Shawn; SULLIVAN Jean; JAYARAMAN Sriram; **SOLI Sigfrid D**

House Ear Inst., 2100 W. 3rd St., Los Angeles, CA 90057

The 127th Meeting of the Acoustical Society of America (Cambridge, Massachusetts (USA)) 1994-06-06/1994-06-10

Journal: Journal of the Acoustical Society of America, 1994-05, 95 (5)  
2991-2991

Language: English

Copyright (c) 1994 American Institute of Physics

**Method for fitting binaural hearing aids**

GAO Shawn; SULLIVAN Jean; JAYARAMAN Sriram; **SOLI Sigfrid D**

For a **hearing aid** wearer to perform binaural sound localization and to utilize directional hearing in noisy environments, it...

... audible levels the binaural cues (i.e., interaural time and level differences) present without the **hearing aid** (s) in place. A method of achieving this **hearing aid** fitting goal for use with a prototype digital signal processing **hearing aid** has been developed. The method includes two major steps: **hearing aid** equalization (HAE) and hearing loss compensation (HLC). HAE is achieved with an FIR filter, which equalizes the amplitude and phase insertion effects of the **hearing aids** and maintains the binaural cues with the **hearing aid** (s) in place. The HAE filter coefficients are obtained from in situ probe tube measures...

... response for the HLC filter is determined from measures of electrical signal levels in the **hearing aid** circuit during threshold tests and during reference signal presentations in the sound field. The HAE...

English Descriptors: Experimental study; Measuring methods; **HEARING AIDS**  
; Sound sources; Noise; Signal processing; Filters; Loudness

June 27, 2003

File 16:Gale Group PROMT(R) 1990-2003/Jun 26  
(c) 2003 The Gale Group  
File 160:Gale Group PROMT(R) 1972-1989  
(c) 1999 The Gale Group  
File 148:Gale Group Trade & Industry DB 1976-2003/Jun 25  
(c)2003 The Gale Group  
File 621:Gale Group New Prod.Annou.(R) 1985-2003/Jun 25  
(c) 2003 The Gale Group  
File 636:Gale Group Newsletter DB(TM) 1987-2003/Jun 24  
(c) 2003 The Gale Group  
File 88:Gale Group Business A.R.T.S. 1976-2003/Jun 24  
(c) 2003 The Gale Group  
File 47:Gale Group Magazine DB(TM) 1959-2003/Jun 23  
(c) 2003 The Gale group  
File 275:Gale Group Computer DB(TM) 1983-2003/Jun 26  
(c) 2003 The Gale Group  
File 570:Gale Group MARS(R) 1984-2003/Jun 26  
(c) 2003 The Gale Group  
File 15:ABI/Inform(R) 1971-2003/Jun 27  
(c) 2003 ProQuest Info&Learning  
File 98:General Sci Abs/Full-Text 1984-2003/May  
(c) 2003 The HW Wilson Co.  
File 674:Computer News Fulltext 1989-2003/Jun W4  
(c) 2003 IDG Communications  
File 9:Business & Industry(R) Jul/1994-2003/Jun 26  
(c) 2003 Resp. DB Svcs.  
File 370:Science 1996-1999/Jul W3  
(c) 1999 AAAS  
File 369:New Scientist 1994-2003/Jun W4  
(c) 2003 Reed Business Information Ltd.  
File 810:Business Wire 1986-1999/Feb 28  
(c) 1999 Business Wire  
File 484:Periodical Abs Plustext 1986-2003/Jun W4  
(c) 2003 ProQuest  
File 647:CMP Computer Fulltext 1988-2003/Jun W1  
(c) 2003 CMP Media, LLC  
File 20:Dialog Global Reporter 1997-2003/Jun 27  
(c) 2003 The Dialog Corp.  
File 696:DIALOG Telecom. Newsletters 1995-2003/Jun 26  
(c) 2003 The Dialog Corp.  
File 634:San Jose Mercury Jun 1985-2003/Jun 26  
(c) 2003 San Jose Mercury News  
File 553:Wilson Bus. Abs. FullText 1982-2003/May  
(c) 2003 The HW Wilson Co  
File 635:Business Dateline(R) 1985-2003/Jun 27  
(c) 2003 ProQuest Info&Learning

Set	Items	Description
S1	977520	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	606149	FILTER?
S3	46195	(BAND? OR HIGH? OR LOW?) ( ) (PASS OR STOP? OR LIMIT?) OR PAS- SBAND?
S4	29710	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	585	S1(3N)S4
S6	0	S5(S)S2(S)S3
S7	0	RD (unique items)
S8	585	S2(5N)S4
S9	0	S8(S)S1(S)S3
S10	0	S9 NOT S7
S11	0	RD (unique items)
S12	3	S1(S)S2(S)S3(S)S4
S13	1	RD (unique items)
S14	1	S13 NOT (S10 OR S7)
S15	4481	S2(3N)ADAPT?

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S16	7	S15(S)S3(S)S4
S17	0	S16(S)S1
S18	0	S17 NOT (S14 OR S10 OR S7)
S19	123	AU=(GAO, S? OR GAO S?)
S20	3	AU=(SOLI, S? OR SOLI S?)
S21	0	AU=(GHI, H? OR GHI H?)
S22	0	S19 AND S20
S23	0	S1(S) (S19 OR S20)

June 27, 2003

14/3,K/1 (Item 1 from file: 88)  
DIALOG(R)File 88:Gale Group Business A.R.T.S.  
(c) 2003 The Gale Group. All rts. reserv.

06352242 SUPPLIER NUMBER: 92939068

**Build the frisker: Sniff out metallic contraband with this hand-held device.**

Sheets, William; Graf, Rudolf F.

Poptronics, 3, 11, 21(1)

Nov, 2002

ISSN: 1526-3681 LANGUAGE: English RECORD TYPE: Fulltext

WORD COUNT: 2820 LINE COUNT: 00206

... at the collector. Components R12, C11, and C12 form a DC-blocking and low-pass **filter** network and suppress the higher frequency components. All we want is the frequency difference product...

...amplifier stage that delivers up to a few hundred milliwatts of audio to a small **speaker** mounted off the PC board.

The Frisker is designed to only sense objects within an...

June 27, 2003

16/3,K/1 (Item 1 from file: 16)  
DIALOG(R)File 16:Gale Group PROMT(R)  
(c) 2003 The Gale Group. All rts. reserv.

02541147 Supplier Number: 43366941 (USE FORMAT 7 FOR FULLTEXT)  
**FOUR-CHIP SET COMBINES FAX, DATA, VOICE: Sierra modem chip set**  
Electronic Engineering Times, p21  
Oct 12, 1992  
Language: English Record Type: Fulltext  
Document Type: Magazine/Journal; Trade  
Word Count: 567

... analog front end includes 12-bit A/D and D/A converters, a seventh-order **low - pass** receive **filter**, and **adaptive** features for near-end **echo cancellation**.  
The SC11083 interface IC sweeps up glue logic for the AT bus, parallel to serial...

16/3,K/2 (Item 1 from file: 148)  
DIALOG(R)File 148:Gale Group Trade & Industry DB  
(c)2003 The Gale Group. All rts. reserv.

03526393 SUPPLIER NUMBER: 06293006 (USE FORMAT 7 OR 9 FOR FULL TEXT)  
**Echo cancellation for high-speed dial-up applications. (part 1 of multipart series) (includes related article on using a V.32 echo cancelling modem) (technical)**  
Turner, Steven E.  
Telecommunications, v22, n1, p80(5)  
Jan, 1988  
DOCUMENT TYPE: technical ISSN: 0278-4831 LANGUAGE: ENGLISH  
RECORD TYPE: FULLTEXT; ABSTRACT  
WORD COUNT: 2381 LINE COUNT: 00182

... the dial-up telephone network.  
\* TYPES OF ECHO  
CANCELERS AND  
HOW THEY WORK  
Data-domain **echo cancelers** can be either baseband or **passband** cancelers or a combination of both. There are two essential differences between baseband and **passband** cancelers. The first is the type of signal (complex and baseband or real and **passband**) contained in the **adaptive** FIR canceler **filter**. The second is the location and technique by which the actual cancellation (subtraction) takes place. A block diagram of a baseband **echo canceler**, as used in a V.32 modem, is shown in Figure 2. A detailed illustration of the actual filter used in the baseband **echo canceler** can be found in Figure 3. Note that the single lines indicate real signals, while...

16/3,K/3 (Item 1 from file: 88)  
DIALOG(R)File 88:Gale Group Business A.R.T.S.  
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04077338 SUPPLIER NUMBER: 18830886  
**Optimum filter banks for signal decomposition and its application in adaptive echo cancellation.**  
Jin, Qu; Luo, Zhi-Quan; Wong, Kon Max  
IEEE Transactions on Signal Processing, v44, n7, p1669(12)  
July, 1996  
ISSN: 1053-587X LANGUAGE: English RECORD TYPE: Abstract

ABSTRACT: The optimum quasi-biorthogonal (QOB) filter banks allow efficient **echo cancellation** in electronic signals. A multiresolution algorithm decomposes a distorted original signal into a number of

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components. An adaptive algorithm **cancels** the **echo** in the received signal. The filters have high energy concentration in the **passband**. The use of adjacent-band **adaptive filtering** along with QQB filters gives better performance than that obtained using in-band filtering alone.

16/3,K/4 (Item 2 from file: 88)  
DIALOG(R)File 88:Gale Group Business A.R.T.S.  
(c) 2003 The Gale Group. All rts. reserv.

03447686 SUPPLIER NUMBER: 15061353  
**Neural networks: applications in industry, business and science.**  
**(Artificial Intelligence) (Cover Story) (Technical)**  
Widrow, Bernard; Rumelhart, David E.; Lehr, Michael A.  
Communications of the ACM, v37, n3, p93(13)  
March, 1994  
DOCUMENT TYPE: Technical ISSN: 0001-0782 LANGUAGE: English  
RECORD TYPE: Fulltext; Abstract  
WORD COUNT: 7127 LINE COUNT: 00713

... lines, which would normally be tolerated with speech, is devastating to high-speed data transmission. **Echo cancelling** solves the problem by detecting the echo and adding an equal and opposite signal to the return path. The cancelling signal is generated by an **adaptive transversal filter** whose coefficients (weights) are automatically adjusted by the LMS algorithm of Widrow and Hoff [32], also known as the delta rule in the field of neural networks. The **adaptive filter** makes use of what amounts to a single neuron. The first **echo cancellers** were developed at AT&T Bell Labs in the 1960s by M. M. Sondhi and...

...fiber-optic channels can have nonflat frequency responses and nonlinear phase responses in the signal **passband**. Sending digital data at high speed through these channels often results in a phenomenon called...

...medium. Equalization in data modems combats this phenomenon by filtering incoming signals. A modem's **adaptive filter**, by **adapting** itself to become a channel inverse, can compensate for the irregularities in channel magnitude and...

16/3,K/5 (Item 1 from file: 275)  
DIALOG(R)File 275:Gale Group Computer DB(TM)  
(c) 2003 The Gale Group. All rts. reserv.

01674631 SUPPLIER NUMBER: 15061353 (USE FORMAT 7 OR 9 FOR FULL TEXT)  
**Neural networks: applications in industry, business and science.**  
**(Artificial Intelligence) (Cover Story) (Technical)**  
Widrow, Bernard; Rumelhart, David E.; Lehr, Michael A.  
Communications of the ACM, v37, n3, p93(13)  
March, 1994  
DOCUMENT TYPE: Technical ISSN: 0001-0782 LANGUAGE: ENGLISH.  
RECORD TYPE: FULLTEXT; ABSTRACT  
WORD COUNT: 8599 LINE COUNT: 00713

... lines, which would normally be tolerated with speech, is devastating to high-speed data transmission. **Echo cancelling** solves the problem by detecting the echo and adding an equal and opposite signal to the return path. The cancelling signal is generated by an **adaptive transversal filter** whose coefficients (weights) are automatically adjusted by the LMS algorithm of Widrow and Hoff [32], also known as the delta rule in the field of neural networks. The **adaptive filter** makes use of what amounts to a single neuron. The first **echo cancellers** were developed at AT&T Bell Labs in the 1960s by M. M. Sondhi and...

...fiber-optic channels can have nonflat frequency responses and nonlinear

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phase responses in the signal **passband** . Sending digital data at high speed through these channels often results in a phenomenon called...

...medium. Equalization in data modems combats this phenomenon by filtering incoming signals. A modem's **adaptive filter** , by **adapting** itself to become a channel inverse, can compensate for the irregularities in channel magnitude and...

16/3,K/6 (Item 1 from file: 15)  
DIALOG(R)File 15:ABI/Inform(R)  
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00389602 88-06435

**Echo Cancellation for High-Speed Dial-Up Applications: Part I**

Turner, Steven E.

Telecommunications v22n1 (North American Edition) PP: 80-87, 104 Jan 1988

ISSN: 0040-2494 JRNL CODE: TEC

ABSTRACT: **Echo canceler** technology has arrived with the advent of the V.32 modem. A critical design issue in the field of data communication, **echo canceler** technology must be understood by design engineers and technical managers. Basically, an **echo canceler** is a filter used to model the echo path created when a transmitted signal flows through the telephone network. Types of **echo cancelers** include data-domain **echo cancelers** , which can be either baseband or **passband** cancelers or a combination of both. Differences between baseband and **passband** cancelers are: 1. the type of signal contained in the **adaptive FIR canceler filter** , and 2. the location and technique by which the actual cancellation takes place. While the baseband canceler requires significant use of complex arithmetic, the **passband** canceler takes much longer to converge and train. The Weinstein canceler is the combined version...  
... a short baseband modeling filter while avoiding the complex arithmetic at the point of actual **echo cancellation** .

16/3,K/7 (Item 1 from file: 647)  
DIALOG(R)File 647:CMP Computer Fulltext  
(c) 2003 CMP Media, LLC. All rts. reserv.

00508707 CMP ACCESSION NUMBER: EET19921012S1572

**FOUR-CHIP SET COMBINES FAX, DATA, VOICE:Sierra modem chip set**

LORING WIRBEL

ELECTRONIC ENGINEERING TIMES, 1992, n 714, 21

PUBLICATION DATE: 921012

JOURNAL CODE: EET LANGUAGE: English

RECORD TYPE: Fulltext

SECTION HEADING: News: Business

WORD COUNT: 567

... analog front end includes 12-bit A/D and D/A converters , a seventh-order **low - pass** receive filter , and **adaptive** features for near-end **echo cancellation** .

The SC11083 interface IC sweeps up glue logic for the AT bus, parallel to serial...

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File 348:EUROPEAN PATENTS 1978-2003/Jun W04

(c) 2003 European Patent Office

File 349:PCT FULLTEXT 1979-2002/UB=20030626,UT=20030619

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Set	Items	Description
S1	6	AU='GAO SHAWN':AU='GAO SHAWN X'
S2	13	AU='SOLI SIGFRID':AU='SOLI SIGFRIED D'
S3	4	AU='GHI'
S4	0	S1 AND S2 AND S3
S5	4	S1 AND S2
S6	0	S1 AND S3

June 27, 2003

5/5,K/1 (Item 1 from file: 348)  
DIALOG(R) File 348:EUROPEAN PATENTS  
(c) 2003 European Patent Office. All rts. reserv.

01154395

**BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS**  
**BANDBEGRENZTE ADAPTIVE RUCKKOPPLUNGSUNTERDRUCKUNG FUR HORHILFEGERATE**  
**DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE**  
**DESTINE AUX PROTHESES AUDITIVES**

PATENT ASSIGNEE:

HOUSE EAR INSTITUTE, (2139940), 5th floor, 21000 West Thirt Street, Los Angeles, CA 90057, (US), (Applicant designated States: all)

INVENTOR:

GAO, Shawn , 18304 Susan Place, Cerritos, CA 91024, (US)

SOLI, Sigfrid , 2020 North Santa Anita Avenue, Sierra Madre, CA 91024, (US)

CHI, Hsiang-Feng, 16907 Larbrook Drive, Hacienda Heights, CA 91745, (US)  
LEGAL REPRESENTATIVE:

Wombwell, Francis (46021), Potts, Kerr & Co. 15, Hamilton Square, Birkenhead Merseyside L41 6BR, (GB)

PATENT (CC, No, Kind, Date): EP 1118247 A2 010725 (Basic)  
WO 200019605 000406

APPLICATION (CC, No, Date): EP 99948516 990930; WO 99US22757 990930

PRIORITY (CC, No, Date): US 102557 P 980930

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI; LU; MC; NL; PT; SE

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: H04R-025/00; H04R-003/02

NOTE:

No A-document published by EPO

LEGAL STATUS (Type, Pub Date, Kind, Text):

Application: 000531 A2 International application. (Art. 158(1))

Application: 000531 A2 International application entering European phase

Application: 010725 A2 Published application without search report

Examination: 010725 A2 Date of request for examination: 20010514

LANGUAGE (Publication,Procedural,Application): English; English; English

INVENTOR:

GAO, Shawn ...

...US)

SOLI, Sigfrid ...

5/5,K/2 (Item 2 from file: 348)  
DIALOG(R) File 348:EUROPEAN PATENTS  
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01037051

**METHOD OF MEASURING AND PREVENTING UNSTABLE FEEDBACK IN HEARING AIDS**  
**PROCEDE DE MESURE ET DE PREVENTION DE RETROACTION INSTABLE DANS DES**  
**PROTHESES AUDITIVES**

PATENT ASSIGNEE:

HOUSE EAR INSTITUTE, (2139940), 5th floor, 21000 West Thirt Street, Los Angeles, CA 90057, (US), (Applicant designated States: all)

INVENTOR:

GAO, Shawn, X. , 5th floor 2100 West Third Street, Los Angeles, CA 90057, (US)

SOLI, Sigfrid, D. , 5th floor 2100 West Third Street, Los Angeles, CA 90057, (US)

PATENT (CC, No, Kind, Date):

WO 9912388 990311

APPLICATION (CC, No, Date): WO 98946842 980903; WO 98US18442 980903

PRIORITY (CC, No, Date): US 926320 970905

June 27, 2003

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;  
LU; MC; NL; PT; SE  
INTERNATIONAL PATENT CLASS: H04R-025/00  
LEGAL STATUS (Type, Pub Date, Kind, Text):  
Application: 010131 A1 International application. (Art. 158(1))  
Application: 990602 A1 International application (Art. 158(1))  
Withdrawal: 010131 A1 Date application deemed withdrawn: 20000406  
Appl Changed: 010131 A1 International application not entering European  
phase  
LANGUAGE (Publication,Procedural,Application): English; English; English

INVENTOR:  
GAO, Shawn, X ...  
...US)  
SOLI, Sigfrid, D ...

5/5,K/3 (Item 1 from file: 349)  
DIALOG(R)File 349:PCT FULLTEXT  
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00556232 \*\*Image available\*\*  
**BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS**  
**DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE**  
**DESTINE AUX PROTHESES AUDITIVES**

Patent Applicant/Assignee:

HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn ,

SOLI Sigfrid ,

CHI Hsiang-Feng

Patent and Priority Information (Country, Number, Date):

Patent: WO 200019605 A2 20000406 (WO 0019605)

Application: WO 99US22757 19990930 (PCT/WO US9922757)

Priority Application: US 98102557 19980930

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ CZ DE  
DE DK DM EE EE ES FI FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP  
KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG  
SI SK SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ  
TZ UG ZW AM AZ BY KG KZ MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE  
IT LU MC NL PT SE BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

International Patent Class: H04R-003/02

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 9255

#### English Abstract

An improved method for adaptively cancelling acoustic feedback in hearing aids and other audio amplification devices. Feedback cancellation is limited to a frequency band that encompasses all unstable frequencies. By limiting the bandwidth of the feedback cancellation signal, the distortion due to the adaptive filter is minimized and limited only to the unstable feedback regions. A relatively simple signal processing algorithm is used to produce highly effective results with minimal signal distortion.

#### French Abstract

L'invention concerne un procede ameliore pour supprimer de maniere adaptative l'effet Larsen dans les protheses auditives et dans d'autres dispositifs audio amplifies. La suppression de l'effet Larsen est limitee a la bande de frequences qui englobe toutes les frequences instables. En

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limitant la bande de frequences du signal d'annulation de l'effet Larsen, on arrive a reduire au minimum la distorsion provoquee par le filtre adaptatif, qui est limitee uniquement aux zones instables de l'effet Larsen. On utilise un algorithme relativement simple de traitement des signaux pour obtenir des resultats probants, et ce avec une distorsion minimale des signaux.

Inventor(s):

GAO Shawn ...

... SOLI Sigfrid

5/5,K/4 (Item 2 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

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00481036 \*\*Image available\*\*

**METHOD OF MEASURING AND PREVENTING UNSTABLE FEEDBACK IN HEARING AIDS**  
**PROCEDE DE MESURE ET DE PREVENTION DE RETROACTION INSTABLE DANS DES**  
**PROTHESES AUDITIVES**

Patent Applicant/Assignee:

HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn X ,

SOLI Sigfrid D

Patent and Priority Information (Country, Number, Date):

Patent: WO 9912388 A1 19990311

Application: WO 98US18442 19980903 (PCT/WO US9818442)

Priority Application: US 97926320 19970905

Designated States: AL AM AT AU AZ BA BB BG BR BY CA CH CN CU CZ DE DE

DK DK EE EE ES FI FI GB GE GH GM HR HU ID IL IS JP KE KG KP KR KZ LC LK

LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL

TJ TM TR TT UA UG UZ VN YU ZW GH GM KE LS MW SD SZ UG ZW AM AZ BY KG KZ

MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE BF BJ

CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 6962

English Abstract

A "true" hearing aid transfer function ( $K(f)$ ), including feedback, is derived from measurements taken with the hearing aid fitted in a patient's ear canal. Closed loop transfer functions ( $L(f)$ ) are calculated at several hearing aid gains without opening the internal circuitry of the hearing aid using a time domain Weiner optimal filter model. The combined open loop transfer function of the hearing aid and feedback path is then calculated. Once the open loop transfer function is known, potentially unstable frequencies are identified and maximum hearing aid gain settings are determined. The hearing aid transfer function and transfer function of feedback path ( $B(f)$ ) are also calculated from the closed loop transfer function measurements.

French Abstract

L'invention concerne une fonction "veritable" de transfert de prothese auditive ( $K(f)$ ), y compris une retroaction, derivee de mesures prises avec la prothese auditive ajustee dans le conduit auditif externe d'un patient. On calcule des fonctions de transfert en boucle fermee ( $L(f)$ ) a plusieurs gains de prothese auditive sans ouvrir l'ensemble de circuits internes de la prothese auditive grace a un modele de filtre optimal de Weiner a reponse temporelle. On calcule ainsi la fonction de transfert en boucle ouverte combinee de la prothese auditive et de la trajectoire de

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retroaction. Une fois que la fonction de transfert en boucle ouverte est connue, on identifie les frequences potentiellement instables et on determine les reglages de gains de prothese auditive maximaux. On calcule egalement la fonction de transfert de la prothese auditive et la fonction de transfert de la trajectoire de retroaction ( $B(f)$ ), a partir des mesures de fonction de transfert en boucle fermee.

Inventor(s):

GAO Shawn X ...

... SOLI Sigfrid D

June 27, 2003

File 344:Chinese Patents Abs Aug 1985-2003/Mar  
(c) 2003 European Patent Office  
File 347:JAPIO Oct 1976-2003/Feb(Updated 030603)  
(c) 2003 JPO & JAPIO  
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200340  
(c) 2003 Thomson Derwent

Set	Items	Description
S1	200	AU='GAO S'
S2	11	AU='SOLI S':AU='SOLI S D'
S3	0	AU='GHI H'
S4	2	S1 AND S2

June 27, 2003

4/5/1 (Item 1 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
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013145713 \*\*Image available\*\*  
WPI Acc No: 2000-317585/200027  
XRPX Acc No: N00-238393

**Adaptive feedback canceller for audio amplification devices e.g. hearing aid, has band limited filter with passband encompassing all unstable frequencies**

Patent Assignee: HOUSE EAR INST (HOUS-N)  
Inventor: CHI H; GAO S; SOLI S  
Number of Countries: 089 Number of Patents: 004  
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200019605	A2	20000406	WO 99US22757	A	19990930	200027 B
AU 9961680	A	20000417	AU 9961680	A	19990930	200035
EP 1118247	A2	20010725	EP 99948516	A	19990930	200143
			WO 99US22757	A	19990930	
JP 2002526961	W	20020820	WO 99US22757	A	19990930	200258
			JP 2000572997	A	19990930	

Priority Applications (No Type Date): US 98102557 P 19980930

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 200019605	A2	E	66	H03H-021/00	
Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW					
Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW					
AU 9961680	A			H03H-021/00	Based on patent WO 200019605
EP 1118247	A2	E		H04R-025/00	Based on patent WO 200019605
Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT LI LT LU LV MC MK NL PT RO SE SI					
JP 2002526961	W		58	H04B-003/23	Based on patent WO 200019605

Abstract (Basic): WO 200019605 A2

NOVELTY - The feedback canceller includes an adaptive digital filter (30) whose output is combined with input of audio amplification device. A band limiting filter having passband limited to a frequency band containing unstable frequencies is coupled between the amplification device and adaptive filter.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for method for adaptively canceling acoustic feedback.

USE - For audio amplification devices such as hearing aid.

ADVANTAGE - By limiting the bandwidth of the feedback cancellation signal, distortion due to adaptive filter is minimized and limited only to unstable feedback regions.

DESCRIPTION OF DRAWING(S) - The figure shows functional block diagram of hearing aid and feedback canceller.

Adaptive digital filter (30)

pp; 66 DwgNo 4/21

Title Terms: ADAPT; FEEDBACK; CANCEL; AUDIO; AMPLIFY; DEVICE; HEARING; AID; BAND; LIMIT; FILTER; PASSBAND; ENCOMPASSING; UNSTABLE; FREQUENCY

Derwent Class: U22; U25

International Patent Class (Main): H04B-003/23; H04R-025/00

International Patent Class (Additional): H03H-021/00; H04R-003/02

File Segment: EPI

4/5/2 (Item 2 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
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009941285    \*\*Image available\*\*

WPI Acc No: 1994-208997/199425

XRFX Acc No: N94-164458

**Signal processing to maintain directional hearing with hearing aid -  
using filter compensating for insertion effect derived from ratio of  
unaided to aided head related transfer function**

Patent Assignee: HOUSE EAR INST (HOUS-N)

Inventor: **GAO S** ; JAYARAMAN S; **SOLI S D** ; SULLIVAN J

Number of Countries: 001    Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 5325436	A	19940628	US 9385652	A	19930630	199425 B

Priority Applications (No Type Date): US 9385652 A 19930630

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
US 5325436	A		17	H04R-005/00	

Abstract (Basic): US 5325436 A

The method for obtaining coefficients of a digital filter for use in compensating effects of a hearing aid involves determining an unaided head related transfer function for each ear and for several azimuth locations of a sound source. An aided head related transfer function is determined for each ear using a hearing aid for the several azimuth locations of the sound source.

Minimum phase representation of the unaided and aided head related transfer function are found. The ratio between the unaided and aided minimum phase representation is calculated to form a target filter response. Several filter coefficients are obtained by sampling the target filter response at several frequency values corresponding to frequency increments in the digital filter.

USE/ADVANTAGE - Allows user to determine direction of sound.

Dwg.2/10

Title Terms: SIGNAL; PROCESS; MAINTAIN; DIRECTION; HEARING; HEARING; AID;  
FILTER; COMPENSATE; INSERT; EFFECT; DERIVATIVE; RATIO; UNAIDED; AID; HEAD  
; RELATED; TRANSFER; FUNCTION

Derwent Class: U22; W04

International Patent Class (Main): H04R-005/00

International Patent Class (Additional): H04R-025/00; H04R-029/00

File Segment: EPI

June 27, 2003

File 344:Chinese Patents Abs Aug 1985-2003/Mar  
(c) 2003 European Patent Office  
File 347:JAPIO Oct 1976-2003/Feb(Updated 030603)  
(c) 2003 JPO & JAPIO  
File 350:Derwent WPIX 1963-2003/UD,UM &UP=200340  
(c) 2003 Thomson Derwent

Set	Items	Description
S1	90144	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	622371	FILTER?
S3	83703	(BAND? OR HIGH? OR LOW?) () (PASS OR STOP? OR LIMIT?) OR PAS- SBAND?
S4	16615	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	33	S1 AND S2 AND S3 AND S4
S6	158	S1(3N)S4
S7	6	S6 AND S2 AND S3
S8	329	S2()ADAPT?
S9	0	S8 AND S1 AND S3 AND S4
S10	148	S3(5N)S4
S11	7	S10 AND S1 AND S2
S12	6	S11 NOT S7
S13	21	S5 NOT (S7 OR S12)
S14	6	S13 AND IC=H04R-025/00
S15	15	S13 NOT S14

June 27, 2003

7/5/1 (Item 1 from file: 347)  
DIALOG(R)File 347:JAPIO  
(c) 2003 JPO & JAPIO. All rts. reserv.

02677118 \*\*Image available\*\*  
LOUD-SPEAKER TELEPHONE SET

PUB. NO.: 63-294018 [JP 63294018 A]  
PUBLISHED: November 30, 1988 (19881130)  
INVENTOR(s): ITO YOSHIO  
MIYAMOTO RYOICHI  
NAKANO YOSHIKAZU  
APPLICANT(s): OKI ELECTRIC IND CO LTD [000029] (A Japanese Company or  
Corporation), JP (Japan)  
APPL. NO.: 62-127935 [JP 87127935]  
FILED: May 27, 1987 (19870527)  
INTL CLASS: [4] H04B-003/23; H04M-001/60; H04M-009/08  
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4  
(COMMUNICATION -- Telephone)  
JOURNAL: Section: E, Section No. 734, Vol. 13, No. 126, Pg. 85, March  
28, 1989 (19890328)

#### ABSTRACT

PURPOSE: To obtain a loud-speaker telephone set for automobile superior in service quality by erasing a passing-round signal only in a certain partial band of the voice signal band for the purpose of preventing the signal, which a speaker in the automobile sends, from passing round to the speaker after a certain time.

CONSTITUTION: A transmission signal Sk from a microphone 3 is inputted to a low - pass filter 14, and only its low band component SLk is taken out and is inputted to an echo canceller 15. An estimated value -rLk of a low band component rLk from a band separating filter 16 is estimated in the canceller 15, and its phase inverted signal is inputted to an adder 17. Consequently, the signal rLk is erased in the output of the adder 17. Its erase error eLk is inputted to a band synthesizing filter 18. A high band component rHk from the filter 16 and the erase error eLk are synthesized into an original voice band signal RDk by the filter 18. This signal is inputted to a speaker 2 and an echo canceller 1. Consequently, the low band component is erased from a passing-round signal rk of the signal inputted from the microphone 3 with respect to the signal RDk.

7/5/2 (Item 1 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
(c) 2003 Thomson Derwent. All rts. reserv.

013145713 \*\*Image available\*\*  
WPI Acc No: 2000-317585/200027  
XRPX Acc No: N00-238393

**Adaptive feedback canceller for audio amplification devices  
e.g. hearing aid, has band limited filter with passband  
encompassing all unstable frequencies**

Patent Assignee: HOUSE EAR INST (HOUS-N)  
Inventor: CHI H; GAO S; SOLI S  
Number of Countries: 089 Number of Patents: 004  
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200019605	A2	20000406	WO 99US22757	A	19990930	200027 B
AU 9961680	A	20000417	AU 9961680	A	19990930	200035
EP 1118247	A2	20010725	EP 99948516	A	19990930	200143
			WO 99US22757	A	19990930	
JP 2002526961	W	20020820	WO 99US22757	A	19990930	200258
			JP 2000572997	A	19990930	

June 27, 2003

Priority Applications (No Type Date): US 98102557 P 19980930

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 200019605 A2 E 66 H03H-021/00

Designated States (National): AE AL AM AT AU AZ BA BB BG BR BY CA CH CN  
CR CU CZ DE DK DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP  
KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG  
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

Designated States (Regional): AT BE CH CY DE DK EA ES FI FR GB GH GM GR  
IE IT KE LS LU MC MW NL OA PT SD SE SL SZ TZ UG ZW

AU 9961680 A H03H-021/00 Based on patent WO 200019605

EP 1118247 A2 E H04R-025/00 Based on patent WO 200019605

Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT  
LI LT LU LV MC MK NL PT RO SE SI

JP 2002526961 W 58 H04B-003/23 Based on patent WO 200019605

Abstract (Basic): WO 200019605 A2

NOVELTY - The feedback canceller includes an adaptive digital  
**filter** (30) whose output is combined with input of audio amplification  
device. A **band limiting filter** having **passband** limited to a  
frequency band containing unstable frequencies is coupled between the  
amplification device and adaptive **filter**.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for  
method for adaptively canceling acoustic feedback.

USE - For audio amplification devices such as hearing aid.

ADVANTAGE - By limiting the bandwidth of the feedback cancellation  
signal, distortion due to adaptive **filter** is minimized and limited  
only to unstable feedback regions.

DESCRIPTION OF DRAWING(S) - The figure shows functional block  
diagram of **hearing aid** and **feedback canceller**.

Adaptive digital **filter** (30)  
pp; 66 DwgNo 4/21

Title Terms: ADAPT; FEEDBACK; CANCEL; AUDIO; AMPLIFY; DEVICE; HEARING; AID;  
BAND; LIMIT; **FILTER**; **PASSBAND**; ENCOMPASSING; UNSTABLE; FREQUENCY

Derwent Class: U22; U25

International Patent Class (Main): H04B-003/23; H04R-025/00

International Patent Class (Additional): H03H-021/00; H04R-003/02

File Segment: EPI

7/5/3 (Item 2 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012858799 \*\*Image available\*\*

WPI Acc No: 2000-030632/200003

XRPX Acc No: N00-023656

**Amplification type intercom apparatus - includes microphones in input  
side of echo canceler and speakers connected in output side of echo  
cancelers in both base-station and sub-station**

Patent Assignee: AIHON KK (AIHO-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 11298618	A	19991029	JP 9899263	A	19980410	200003 B

Priority Applications (No Type Date): JP 9899263 A 19980410

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

JP 11298618 A 12 H04M-009/08

Abstract (Basic): JP 11298618 A

NOVELTY - Echo cancelers (M15,T15) of base-station (M1) and  
sub-station (T1) are connected to the output side of **high pass  
filters** (M22,T22), respectively. Microphones (M12,T12) are connected

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to input of each **echo canceler**, and **speakers** (M17,T17) are connected to the output of echo cancelers. **High pass filters** (M22,T22) are connected to input of codec (M16,T16) connected to the line (L1).

USE - For e.g. hands-free speaker phone.

ADVANTAGE - Enables transmission and reception of more audio signals having high frequency. DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of intercom apparatus. (L1) Line; (M1) Base-station; (M12,T12) Microphones; (M15,T15) Echo cancelers; (M16,T16) Codec; (M17,T17) Speakers; (M22,T22) **High pass filters**; (T1) Sub-station.

Dwg.1/3

Title Terms: AMPLIFY; TYPE; INTERCOMMUNICATION; APPARATUS; MICROPHONE; INPUT; SIDE; ECHO; SPEAKER; CONNECT; OUTPUT; SIDE; ECHO; BASE; STATION; SUB; STATION

Derwent Class: W01

International Patent Class (Main): H04M-009/08

International Patent Class (Additional): H04M-001/60

File Segment: EPI

7/5/4 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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010974026 \*\*Image available\*\*

WPI Acc No: 1996-470975/199647

XRFX Acc No: N96-397198

**Howling canceler for preventing feedback of sound from speaker to microphone - has digital-analog converter which performs digital-analog conversion of each frequency-band signal after concerned signal has been controlled to stable side**

Patent Assignee: JAPAN RADIO CO LTD (NIUR )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 8237789	A	19960913	JP 9558263	A	19950223	199647 B

Priority Applications (No Type Date): JP 9558263 A 19950223

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 8237789	A		3	H04R-003/02	

Abstract (Basic): JP 8237789 A

The canceler includes an analog-digital converter (2) which converts an audio signal input to a microphone to a digital signal. The obtd. digital signal is divided into several frequency-band signal of desired number of channels through a **band - pass filter**. An oscillation sensor detects the existence of oscillation at each frequency-band signal.

An adder (6) combines the frequency band signals that passes to an oscillation prevention circuit. A digital-analog converter (7) performs the digital-analog conversion of each frequency-band signal after the concerned signal has been controlled to a stable side.

ADVANTAGE - Offers howling canceler which pertinently prevents howling during signal processing. Prevents changing of microphone direction and specific speaker interruption.

Dwg.1/2

Title Terms: HOWLING; PREVENT; FEEDBACK; SOUND; SPEAKER; MICROPHONE; DIGITAL; ANALOGUE; CONVERTER; PERFORMANCE; DIGITAL; ANALOGUE; CONVERT; FREQUENCY; BAND; SIGNAL; AFTER; CONCERN; SIGNAL; CONTROL; STABILISED; SIDE

Derwent Class: U25; W04

International Patent Class (Main): H04R-003/02

International Patent Class (Additional): H03G-005/14

June 27, 2003

File Segment: EPI

7/5/5 (Item 4 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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010463052 \*\*Image available\*\*

WPI Acc No: 1995-364371/199547

XRPX Acc No: N95-269538

**Loud-speaker terminal equipment - has alarm tone transmission function  
corresp. to volume approved by fire-fighting authority with speaker  
built with acoustic feedback loop between microphones**

Patent Assignee: AIHON KK (AIHO-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7248789	A	19950926	JP 9442489	A	19940314	199547 B

Priority Applications (No Type Date): JP 9442489 A 19940314

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 7248789	A	4	G10K-015/04	

Abstract (Basic): JP 7248789 A

The equipment includes a microphone (1) that generates a specific frequency signal (f2) from an audio signal (f1) that passes through a **band - pass filter** (3).

An electronic switch (4) is activated according to an alarm signal (f6) produced by a sensor (19) in passing the specific frequency signal into a speaker (10) which is built with an acoustic-feedback loop between the microphones.

**ADVANTAGE** - Enables reduction of electric audio conversion efficiency caused by change in manufacturing error of speaker without enlarging power amplifier that drives it.

Dwg.1/3

Title Terms: LOUD; SPEAKER; TERMINAL; EQUIPMENT; ALARM; TONE; TRANSMISSION; FUNCTION; CORRESPOND; VOLUME; APPROVE; FIRE; FIGHTING; AUTHORISE; SPEAKER ; BUILD; ACOUSTIC; FEEDBACK; LOOP; MICROPHONE

Derwent Class: P86; W01; W05

International Patent Class (Main): G10K-015/04

International Patent Class (Additional): H04M-011/04; H04R-003/04

File Segment: EPI; EngPI

7/5/6 (Item 5 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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007529265 \*\*Image available\*\*

WPI Acc No: 1988-163197/198824

XRPX Acc No: N88-124686

**Super-regenerative detector with saw device e.g. for portable phone - has  
quench oscillator to switch RF oscillator including SAW device between  
oscillation and non-oscillation**

Patent Assignee: RF MONOLITHICS INC (RFMO-N)

Inventor: ASH D L

Number of Countries: 007 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 271190	A	19880615	EP 87308927	A	19871008	198824 B
US 4749964	A	19880607	US 86939527	A	19861208	198825
JP 63198404	A	19880817	JP 87304646	A	19871203	198839
EP 271190	B1	19940302	EP 87308927	A	19871008	199409
DE 3789206	G	19940407	DE 3789206	A	19871008	199415

June 27, 2003

EP 87308927      A      19871008

Priority Applications (No Type Date): US 86939527 A 19861208

Cited Patents: 1.Jnl.Ref; A3...8922; EP 184508; FR 2209255; No-SR.Pub; US 3119065; US 3405364; US 4143324

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
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EP 271190	A	E	39		
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Designated States (Regional): DE FR GB IT NL

US 4749964	A		8		
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EP 271190	B1	E	9	H03D-011/04	
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Designated States (Regional): DE FR GB IT NL

DE 3789206	G			H03D-011/04	Based on patent EP 271190
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Abstract (Basic): EP 271190 A

The detector circuit uses a single transistor (Q1) to the collector of which the modulated RF signal is coupled from a terminal terminal (20) through a coupling capacitor (C5). The RF signal is coupled also through a **feedback loop** including the surface **acoustic wave device** (22) and an inductor (L2) to initiate oscillations more rapidly than is the case with thermal noise alone as the input voltage.

The SAW device has a relatively low quality factor and low loss. The transistor output is coupled through an inductor (L1) and capacitor (C3) to a **low pass filter** (24) for recovery of the modulation signal.

USE/ADVANTAGE - In garage door opening receiver. Is temperature stable, does not drift in frequency, and has very narrow reception band eliminating effects of noise and stray signals

Title Terms: SUPER; REGENERATE; DETECT; SAW; DEVICE; PORTABLE; TELEPHONE; QUENCH; OSCILLATOR; SWITCH; RF; OSCILLATOR; SAW; DEVICE; OSCILLATING; NON ; OSCILLATING

Index Terms/Additional Words: SUPER; REGENERATE; DETECT; SAW; DEVICE; PORTABLE

Derwent Class: U14; U23; W01; W02; W05; X25

International Patent Class (Main): H03D-011/04

International Patent Class (Additional): H03B-005/00; H03D-001/18

File Segment: EPI

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12/5/1 (Item 1 from file: 347)  
DIALOG(R)File 347:JAPIO  
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06357010 \*\*Image available\*\*  
LOUD **SPEAKER** INTERPHONE SYSTEM

PUB. NO.: 11-298618 [JP 11298618 A]  
PUBLISHED: October 29, 1999 (19991029)  
INVENTOR(s): NISHIMURA TOMOHIRO  
KITAGAWA KAZUMI  
APPLICANT(s): AIPHONE CO LTD  
APPL. NO.: 10-099263 [JP 9899263]  
FILED: April 10, 1998 (19980410)  
INTL CLASS: H04M-009/08; H04M-001/60

#### ABSTRACT

PROBLEM TO BE SOLVED: To transmit/receive audio signals at a much higher frequency without being limited to audio signals in a telephone **band limited** by an **echo canceler** with respect to a loud **speaker** interphone system.

SOLUTION: Among audio signals exchanged between a base unit M1 and a hand set T1 connected through a line L1, audio signals in the telephone band are respectively discharged from first **speakers** M11 and T11 through first and third band-pass **filters** M21 T21, M23 and T23 and echo cancelers M15 and T15 and audio signals at the frequency higher than the telephone band are respectively discharged from second **speakers** M17 and T17 separately provided under the control of audio switches M19 and T19 through second and fourth band-pass switches M22, T22, M24 and T24. Thus, the audio signals in the telephone band can be made into fully duplex communication and the audio signals at the frequency higher than the telephone band can be made into semi-duplex communication.

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12/5/2 (Item 2 from file: 347)  
DIALOG(R)File 347:JAPIO  
(c) 2003 JPO & JAPIO. All rts. reserv.

05449992 \*\*Image available\*\*  
LOUD **SPEAKER** INFORMATION COMMUNICATION SYSTEM

PUB. NO.: 09-064792 [JP 9064792 A]  
PUBLISHED: March 07, 1997 (19970307)  
INVENTOR(s): IWASAKI TAKASHI  
KUSANO YOSHIMASA  
APPLICANT(s): KYOCERA CORP [358923] (A Japanese Company or Corporation), JP (Japan)  
APPL. NO.: 07-210495 [JP 95210495]  
FILED: August 18, 1995 (19950818)  
INTL CLASS: [6] H04B-003/23; H03H-021/00; H04M-001/60  
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 33.1 (MARINE DEVELOPMENT -- Space Utilization); 34.4 (SPACE DEVELOPMENT -- Communication); 41.5 (MATERIALS -- Electric Wires & Cables); 44.1 (COMMUNICATION -- Transmission Circuits & Antennae); 44.4 (COMMUNICATION -- Telephone); 44.6 (COMMUNICATION -- Television)

#### ABSTRACT

PROBLEM TO BE SOLVED: To have excellent high speed and operation stability and high adaptive performance and to enable an acoustic control, always maintaining large acoustic **echo canceling** amount by inserting a **low-pass filter** into a transmitting signal output terminal and interrupting

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the high frequency component of a transmitting signal when howling is detected.

SOLUTION: The device having the same constitution as an **acoustic** echo removing **device** 11 adopting a learning identifying method as an adaptive algorithm is composed of a howling detector 12, a band limit controller, a changeover switch 15 and a low-pass **filter** 16. When howling is detected by this howling detector 12, the band limit controller stops the successive update operation of a coefficient correction amount arithmetic circuit 7, operates the changeover switch 15, inserts the low-pass **filter** 16 and interrupts the high frequency component of a transmitting signal. Therefore, even if a communication line state is fallen into an unstable status, the generation of howling is suppressed without disconnecting the line and a speech state which is excellent in high speed and operation stability can be maintained.

12/5/3 (Item 3 from file: 347)

DIALOG(R)File 347:JAPIO

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02731494 \*\*Image available\*\*

ECHO ERASING DEVICE

PUB. NO.: 01-029094 [JP 1029094 A]

PUBLISHED: January 31, 1989 (19890131)

INVENTOR(s): OIKAWA HIROSHI

MAKINO SHOJI

MINAMI SHIGENOBU

SAEKI TAKASHI

APPLICANT(s): NIPPON TELEGR & TELEPH CORP <NTT> [000422] (A Japanese

Company or Corporation), JP (Japan)

TOSHIBA CORP [000307] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 62-183611 [JP 87183611]

FILED: July 24, 1987 (19870724)

INTL CLASS: [4] H04R-003/02; H04B-003/23; H04M-009/08

JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment); 44.2 (COMMUNICATION -- Transmission Systems); 44.4 (COMMUNICATION -- Telephone)

JOURNAL: Section: E, Section No. 760, Vol. 13, No. 216, Pg. 163, May 19, 1989 (19890519)

#### ABSTRACT

PURPOSE: To effectively cancel an echo signal by assuming the reverberation time of a echo path from echo path characteristic assumed at each frequency band and variably setting a tap length in order to generate a pseudo echo signal in accordance with it.

CONSTITUTION: A **speaker** 1 and a microphone 2 are acoustically coupled. An input sound signal is separated by low-pass **filters** 11 and 21 and high-pass **filters** 12 and 22, the signals of respective frequency bands are sampled by frequency converting circuits 13, 23, 14 and 24, and the signals are supplied for the generation of the pseudo echo signal by an **echo canceler** circuit 3 for a **low - pass** and an **echo canceler** circuit 4 for a **high - pass**. The pseudo echo signal generated at the **echo canceler** circuit 3 for the **low - pass** and the **echo canceler** circuit 4 for the **high - pass** is supplied for the **canceling** processing of the **echo** signal at subtractors 5 and 6.

12/5/4 (Item 4 from file: 347)

DIALOG(R)File 347:JAPIO

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02313329 \*\*Image available\*\*

ECHO ELIMINATOR

June 27, 2003

PUB. NO.: 62-230229 [JP 62230229 A]  
PUBLISHED: October 08, 1987 (19871008)  
INVENTOR(s): MINAMI SHIGENOBU  
APPLICANT(s): TOSHIBA CORP [000307] (A Japanese Company or Corporation), JP  
(Japan)  
APPL. NO.: 61-073411 [JP 8673411]  
FILED: March 31, 1986 (19860331)  
INTL CLASS: [4] H04B-003/20; H04M-009/08  
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4  
(COMMUNICATION -- Telephone)  
JOURNAL: Section: E, Section No. 594, Vol. 12, No. 99, Pg. 140, March  
31, 1988 (19880331)

#### ABSTRACT

PURPOSE: To eliminate noise produced around the boundary of each band by dividing the band into plural numbers, converting each band into a low frequency, eliminating an echo signal, restoring the frequency into the original band so as to eliminate the echo signal of all bands.

CONSTITUTION: When a transmission/reception discrimination circuit 9 discriminates the mode as the reception mode, the circuit 9 throws a switch 11 to the position B and throws a switch 13 to the position D. As a result, a reception signal inputted from a terminal 23 is outputted from a **speaker** 17 via a **band stop filter** 5, a band split **echo canceller** 3 and an amplifier 15. In this case, the band stop **filter** 5 eliminates the frequency component of the band near the boundary of the band to be split by the band split echo canceller 3, then the noise around it is eliminated by using a quadrature mirror **filter**. The noise is eliminated almost similarly in case of the transmission.

12/5/5 (Item 1 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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014819847 \*\*Image available\*\*

WPI Acc No: 2002-640553/200269

XPX Acc No: N02-506457

**Echo canceller for full-duplex communication device e.g. modem, has sub-band echo canceller which suppresses band limited pseudo echo signal, in addition to adaptive type filter**

Patent Assignee: RICOH KK (RICO )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 2002232329	A	20020816	JP 200121979	A	20010130	200269 B

Priority Applications (No Type Date): JP 200121979 A 20010130

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 2002232329	A		7	H04B-003/23	

Abstract (Basic): JP 2002232329 A

NOVELTY - An adaptive type **filter** suppresses a pseudo echo signal, using transmitted signal and received echo signal. A sub-band **echo canceller** (15) suppresses the **band limited** pseudo echo signal.

USE - In full-duplex communication device e.g. modem, **speaker** phone.

ADVANTAGE - Since band limited pseudo echo signals is suppressed, problem of frequency resolution is avoided.

DESCRIPTION OF DRAWING(S) - The figure shows the block diagram of the echo canceller. (Drawing includes non-English language text).

Sub-band echo canceller (15)

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pp; 7 DwgNo 1/5  
Title Terms: ECHO; CANCEL; FULL; DUPLEX; COMMUNICATE; DEVICE; MODEM; SUB;  
BAND; ECHO; CANCEL; SUPPRESS; BAND; LIMIT; PSEUDO; ECHO; SIGNAL; ADD;  
ADAPT; TYPE; **FILTER**  
Derwent Class: W02  
International Patent Class (Main): H04B-003/23  
File Segment: EPI

12/5/6 (Item 2 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
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010309586 \*\*Image available\*\*  
WPI Acc No: 1995-210844/199528  
XRPX Acc No: N95-165472

**Echo canceller for teleconferencing system - has band division type low pass echo canceller that eliminates remaining echo in adaptive filter after pseudo echo derived from decimeter low pass data is subtracted from second**

Patent Assignee: RICOH KK (RICO )  
Number of Countries: 001 Number of Patents: 001  
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7123028	A	19950512	JP 93265305	A	19931025	199528 B

Priority Applications (No Type Date): JP 93265305 A 19931025

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
JP 7123028	A	7	H04B-003/23	

Abstract (Basic): JP 7123028 A

The echo canceller has an adaptive **filter** which uses two decimeters (5.1,6.1) to generate low pass data. The voice signal received through the voice signal noise correcting microphone (12) is amplified, digitised and modulated with twice the sampling frequency before it is fed to the low frequency echo canceller (8) operating in the 7 KHz band. The echo canceller has a 3.4 KHz sampling frequency.

The digital voice data is transmitted through the codec (2) and the communication controller (2). The digital voice data received as a response is received by the same communication controller. The operation is reversed and the processed voice data is amplified to the **speaker**.

**ADVANTAGE** - Prevents echo generation. Eliminates howling. Increases device versatility since it can be used for both wideband and narrowband communications.

Dwg.1/2

Title Terms: ECHO; CANCEL; TELECONFERENCE; SYSTEM; BAND; DIVIDE; TYPE; LOW; PASS; ECHO; CANCEL; ELIMINATE; REMAINING; ECHO; ADAPT; **FILTER** ; AFTER; PSEUDO; ECHO; DERIVATIVE; LOW; PASS; DATA; SUBTRACT; SECOND

Index Terms/Additional Words: ECHO; CANCEL; TELECONFEREN

Derwent Class: W01; W02

International Patent Class (Main): H04B-003/23

International Patent Class (Additional): H04M-009/08

File Segment: EPI

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14/5/1 (Item 1 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
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013980443 \*\*Image available\*\*  
WPI Acc No: 2001-464657/200150  
Related WPI Acc No: 2001-343016  
XRPX Acc No: N01-344652

Feedback canceling method for acoustic system, involves using least mean square algorithm for generating filter coefficients and providing low frequency input for LMS algorithm

Patent Assignee: OTICON AS (OTIC-N)  
Inventor: EKELID M; NIELSEN J  
Number of Countries: 024 Number of Patents: 003  
Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
WO 200106812	A1	20010125	WO 2000DK380	A	20000707	200150 B
AU 200058064	A	20010205	AU 200058064	A	20000707	200150
EP 1203510	A1	20020508	EP 2000943695	A	20000707	200238
			WO 2000DK380	A	20000707	

Priority Applications (No Type Date): DK 991043 A 19990719

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
WO 200106812	A1	E	23	H04R-025/00	
Designated States (National): AU BR CA JP US					
Designated States (Regional): AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE					
AU 200058064	A			G10L-021/02	Based on patent WO 200106812
EP 1203510	A1	E		H04R-025/00	Based on patent WO 200106812
Designated States (Regional): AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU MC NL PT SE					

Abstract (Basic): WO 200106812 A1

NOVELTY - The **feedback canceling** method involves using LMS (8) algorithm for generating **filter** coefficients (9) and using a **high pass filter** (20) to prevent low frequency signals from entering the LMS algorithm. The low frequency input for the LMS algorithm is provided by using the additional **feedback cancellation filter** (7) and a noise generator.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is also included for **hearing aid**.

USE - For acoustic system to **cancel feedback**.

ADVANTAGE - The method improves adaptation speed and eliminates side effects by fast suppression of feedback oscillations. The user comfort is improved by stabilizing the **feedback cancellation** and providing reliable coefficients for **feedback canceling filter**.

DESCRIPTION OF DRAWING(S) - The figure shows the schematic diagram of **feedback canceling** system.

Additional **feedback cancellation filter** (7)

LMS (8)

**Filter** coefficients (9)

**High pass filter** (20)

pp; 23 DwgNo 1/3

Title Terms: FEEDBACK; METHOD; ACOUSTIC; SYSTEM; MEAN; SQUARE; ALGORITHM; GENERATE; **FILTER**; COEFFICIENT; LOW; FREQUENCY; INPUT; ALGORITHM

Derwent Class: P86; W04

International Patent Class (Main): G10L-021/02; H04R-025/00

File Segment: EPI; EngPI

14/5/2 (Item 2 from file: 350)  
DIALOG(R)File 350:Derwent WPIX  
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008689451

WPI Acc No: 1991-193471/199126

XRFX Acc No: N91-148132

**Programmable hybrid hearing aid with signal processing - has open connection which constitute acoustic transmission channel with low - pass characteristic and resonant amplification**

Patent Assignee: NHA AS (NHAN-N)

Inventor: KROKSTAD A; RAMSTAD T A; SVEAN J; RAMSTAD T

Number of Countries: 033 Number of Patents: 013

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
WO 9108654	A	19910613				199126	B
NO 8904806	A	19910531				199131	
AU 9168805	A	19910626				199139	
FI 9202408	A	19920526	WO 90N0178	A	19901129	199235	
			FI 922408	A	19920526		
EP 502073	A1	19920909	WO 90N0178	A	19901129	199237	
			EP 91900061	A	19901129		
JP 5504029	W	19930624	WO 90N0178	A	19901129	199330	
			JP 91500704	A	19901129		
HU 63726	T	19930928	WO 90N0178	A	19901129	199344	
			HU 921417	A	19901129		
US 5276739	A	19940104	WO 90N0178	A	19901129	199402	
			US 92852242	A	19920526		
EP 502073	B1	19940914	WO 90N0178	A	19901129	199435	
			EP 91900061	A	19901129		
DE 69012582	E	19941020	DE 612582	A	19901129	199441	
			WO 90N0178	A	19901129		
			EP 91900061	A	19901129		
AU 654266	B	19941103	AU 9168805	A	19901129	199501	
ES 2060345	T3	19941116	EP 91900061	A	19901129	199501	
CA 2069737	C	19990914	CA 2069737	A	19901129	200004	
			WO 90N0178	A	19901129		

Priority Applications (No Type Date): NO 894806 A 19891130

Cited Patents: 00 32690500; 00 33554200; 00 36403700; 00 4025900; 4187413

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

WO 9108654 A  
Designated States (National): AT AU BB BG BR CA CH DE DK ES FI GB HU JP  
KP KR LK LU MC MG MW NO RO SD SE SU  
Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LU NL SE US  
CA 2069737 C E H04R-025/00 Based on patent WO 9108654  
EP 502073 A1 E 10 H04R-025/00 Based on patent WO 9108654  
Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LI LU NL SE  
JP 5504029 W H04R-025/00 Based on patent WO 9108654  
HU 63726 T H04R-025/00 Based on patent WO 9108654  
US 5276739 A 19 H04R-025/00 Based on patent WO 9108654  
EP 502073 B1 E 35 H04R-025/00 Based on patent WO 9108654  
Designated States (Regional): AT BE CH DE DK ES FR GB GR IT LI LU NL SE  
DE 69012582 E H04R-025/00 Based on patent EP 502073  
Based on patent WO 9108654  
AU 654266 B H04R-025/00 Previous Publ. patent AU 9168805  
Based on patent WO 9108654  
ES 2060345 T3 H04R-025/00 Based on patent EP 502073  
FI 9202408 A H04R

Abstract (Basic): WO 9108654 A

The programmable hearing aid comprises a main section (1) and two secondary sections (2a,2b) which are connected to the main section together with a battery.

The open connection of the aid constitutes an acoustic transmission channel with low - pass characteristic and resonant amplification . The hearing aid also includes an analog input

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section with a microphone amplifier (11) and a deconvolution **filter** (13). A digital signal processor with a compressor (33) and an equaliser (34) are also provided each of which contains RAM memories together with an analog output section with a reconstruction **filter** (14).

USE/ADVANTAGE - **Hearing aid** permits utilisation of hearing residuee in bass range, and user can choose one of different response functions stored in **hearing aid** according to **acoustic** environment. (Dwg.No. 1a/6)

Title Terms: PROGRAM; HYBRID; HEARING; AID; SIGNAL; PROCESS; OPEN; CONNECT; CONSTITUTE; ACOUSTIC; TRANSMISSION; CHANNEL; LOW; PASS; CHARACTERISTIC; RESONANCE; AMPLIFY

Derwent Class: W04

International Patent Class (Main): H04R-007/185; **H04R-025/00**

International Patent Class (Additional): H04R-025/02

File Segment: EPI

14/5/3 (Item 3 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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008230324 \*\*Image available\*\*

WPI Acc No: 1990-117325/199016

XRPX Acc No: N90-090953

**Integrated compression amplifier with programmable threshold voltage - has feedback circuit containing rectifier and low - pass filter connected to control input of 2-quadrant amplifier**

Patent Assignee: SIEMENS AG (SIEI )

Inventor: MAUTHE M

Number of Countries: 012 Number of Patents: 007

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 363714	A	19900418	EP 89117648	A	19890925	199016 B
CA 2000434	A	19900413				199019
JP 2149111	A	19900607	JP 89266057	A	19891011	199029
US 4987383	A	19910122	US 89412166	A	19890925	199106
CA 2000434	C	19931207	CA 2000434	A	19891011	199404
EP 363714	B1	19950405	EP 89117648	A	19890925	199518
DE 58909157	G	19950511	DE 509157	A	19890925	199524
			EP 89117648	A	19890925	

Priority Applications (No Type Date): DE 3834928 A 19881013

Cited Patents: 3.Jnl.Ref; A3...9041; NoSR.Pub; US 3919654; US 4539440

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes

EP 363714 A  
Designated States (Regional): AT CH DE FR GB IT LI NL

EP 363714 B1 G 32 H03G-009/02  
Designated States (Regional): AT CH DE FR GB IT LI NL

DE 58909157 G H03G-009/02 Based on patent EP 363714

CA 2000434 C H03F-001/38

Abstract (Basic): EP 363714 A

The amplifier has a 2-quadrant multiplier (2M) and a **feedback loop** containing a rectifier stage (GR) and a **low - pass filter** (TP) providing an output signal which is supplied to the control input of the 2-quadrant multiplier (2M). The output of the latter is coupled via a separation amplifier (TV) to a controlled amplification stage (SC).

The rectifier stage (GR) is coupled to a bias voltage generator (GV), with the output of the rectifier stage (GR) coupled to the **low - pass filter** (TP) via a summation point (S) receiving an output current (IO) from a reference current source (RI).

USE - For multi-channel automatic gain controller in **hearing aid**

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. (22pp Dwg.No.4/9)

Title Terms: INTEGRATE; COMPRESS; AMPLIFY; PROGRAM; THRESHOLD; VOLTAGE;  
FEEDBACK; CIRCUIT; CONTAIN; RECTIFY; LOW; PASS; **FILTER** ; CONNECT;  
CONTROL; INPUT; QUADRANT; AMPLIFY

Derwent Class: U24

International Patent Class (Main): H03F-001/38; H03G-009/02

International Patent Class (Additional): H03G-003/12; H03G-007/08;  
H03G-009/12; **H04R-025/00**

File Segment: EPI

**14/5/4 (Item 4 from file: 350)**

DIALOG(R)File 350:Derwent WPIX

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008213435 \*\*Image available\*\*

WPI Acc No: 1990-100436/199014

XRPX Acc No: N90-077626

Hearing aid system - has feedback loop to control variable  
filter to reduce effect of noise

Patent Assignee: BELTONE ELECTRONICS CORP (BELT-N)

Inventor: ANDERSON J R

Number of Countries: 004 Number of Patents: 005

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
DE 3927765	A	19900329	DE 3927765	A	19890823	199014 B
FR 2635680	A	19900302				199016
JP 2113698	A	19900425	JP 89221887	A	19890830	199023
US 5170434	A	19921208	US 88238207	A	19880830	199252
			US 89459309	A	19891229	
			US 91722926	A	19910628	
DE 3927765	C2	19930527	DE 3927765	A	19890823	199321

Priority Applications (No Type Date): US 88238207 A 19880830; US 89459309 A  
19891229; US 91722926 A 19910628

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
DE 3927765	A		8		
US 5170434	A		6	H04R-025/00	Cont of application US 88238207 Cont of application US 89459309
DE 3927765	C2		8	H04R-025/00	

Abstract (Basic): DE 3927765 A

A **hearing aid** has a microphone (12), a variable **filter** (14), amplifier (16) and a sensor unit (18). The microphone receives speech input that is passed through the **filter** which is a **high pass** device with the cut off frequency determined by the control input (20). The sensor circuit has a threshold level control (25), **band pass filter** (26), level detector (30) and a smoothing circuit (32).

The **band pass filter** has a centre frequency of 250 Hz and generates an output that is interpreted by the detector to identify noise. This results in the variable **filter** (14) being adjusted to effectively reduce the noise effect.

ADVANTAGE - Modifies signal to reduce received noise effect

Title Terms: HEARING; AID; SYSTEM; FEEDBACK; LOOP; CONTROL; VARIABLE;  
**FILTER** ; REDUCE; EFFECT; NOISE

Derwent Class: P32; U25; W04

International Patent Class (Main): **H04R-025/00**

International Patent Class (Additional): A61F-011/04

File Segment: EPI; EngPI

**14/5/5 (Item 5 from file: 350)**

DIALOG(R)File 350:Derwent WPIX

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003763649

WPI Acc No: 1983-759861/198337

XRPX Acc No: N83-160439

**Amplifier with automatic gain control for hearing aid - has variable resistor and two transistors in feedback loop**

Patent Assignee: AUDIBEL (AUDI-N); PHILIPS GLOEILAMPENFAB NV (PHIG )

Inventor: RIDEL P

Number of Countries: 007 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
DE 3306441	A	19830908				198337	B
FR 2522451	A	19830902				198340	
GB 2117200	A	19831005				198340	
JP 58162115	A	19830926				198344	
DK 8300939	A	19831107				198351	
US 4509022	A	19850402	US 83470745	A	19830228	198516	
GB 2117200	B	19851211				198550	
IT 1167626	B	19870513				198941	
DE 3306441	C	19910822				199134	

Priority Applications (No Type Date): FR 823347 A 19820301

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
DE 3306441	A	8		

Abstract (Basic): DE 3306441 A

The amplifier has an input transducer, an amplifying unit (30) an output transducer and automatic gain control. A variable resistor (40) is used to obtain an ac voltage having the same phase as the signal from the input transducer. A rectifier (50) and an RC **filter** are provided.

The **filter** 's output is connected to the base of a first transistor (70) whose collector controls the base of a second transistor (80). The second transistor is connected to the input of the amplifier unit such that it increases the short circuiting experienced by this unit's input signal as the signal from the variable resistor becomes larger. The advantage lies in the amplifier's being suitable for the small supply voltages (c. 1.3V) found in **hearing aids**.

1/1

Title Terms: AMPLIFY; AUTOMATIC; GAIN; CONTROL; HEARING; AID; VARIABLE; RESISTOR; TWO; TRANSISTOR; FEEDBACK; LOOP

Derwent Class: U24; W04

International Patent Class (Additional): H03G-003/20; H03G-007/06;

**H04R-025/00**

File Segment: EPI

**14/5/6 (Item 6 from file: 350)**

DIALOG(R)File 350:Derwent WPIX

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003613523

WPI Acc No: 1983-G1722K/198319

XRPX Acc No: N83-080160

**hearing aid amplifier circuit - has filter with wide-band and high - pass filter paths**

Patent Assignee: SIEMENS AG (SIEI )

Inventor: SCHLOSSER H

Number of Countries: 001 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
DE 3141420	A	19830505	DE 3141420	A	19811019	198319	B
DE 3141420	C	19890202				198905	

June 27, 2003

Priority Applications (No Type Date): DE 3141420 A 19811019

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
DE 3141420	A		16		

Abstract (Basic): DE 3141420 A

The amplifier circuit (6,7) is inserted between a microphone and a hearing capsule, for amplifying the received sound waves. It incorporates a volume control and a frequency **filter** (7), the latter exhibiting two signal paths (7a,7b) with wideband and **high pass filter** characteristics respectively. The signals fed along the two signal paths (7a,7b) exhibiting a relative phase shift of about 180 degrees.

The wideband signal path (7a) has a variable resistor (24) and contains only passive components so that it can transmit signals in either direction for simultaneously acting as a **feedback loop**. The other signal path (7b) has a positive amplification of between 5 and 10dB. The amplifier circuit allows optimum sound balancing.

1/10

Title Terms: HEARING; AID; AMPLIFY; CIRCUIT; **FILTER** ; WIDE; BAND; HIGH; PASS; **FILTER** ; PATH

Derwent Class: U25; W04

International Patent Class (Additional): H04R-003/04; **H04R-025/00**

File Segment: EPI

June 27, 2003

15/5/1 (Item 1 from file: 347)  
DIALOG(R)File 347:JAPIO  
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05449993 \*\*Image available\*\*  
LOUD **SPEAKER** INFORMATION COMMUNICATION SYSTEM

PUB. NO.: 09-064793 [JP 9064793 A]  
PUBLISHED: March 07, 1997 (19970307)  
INVENTOR(s): IWASAKI TAKASHI  
KUSANO YOSHIMASA  
APPLICANT(s): KYOCERA CORP [358923] (A Japanese Company or Corporation), JP  
(Japan)  
APPL. NO.: 07-211752 [JP 95211752]  
FILED: August 21, 1995 (19950821)  
INTL CLASS: [6] H04B-003/23; H03H-021/00; H04M-001/60  
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 33.1 (MARINE  
DEVELOPMENT -- Space Utilization); 34.4 (SPACE DEVELOPMENT --  
Communication); 41.5 (MATERIALS -- Electric Wires & Cables);  
44.1 (COMMUNICATION -- Transmission Circuits & Antennae);  
44.4 (COMMUNICATION -- Telephone); 44.6 (COMMUNICATION ---  
Television)

#### ABSTRACT

PROBLEM TO BE SOLVED: To have excellent high speed and operation stability and high adaptive performance and to enable an acoustic control, always maintaining large acoustic **echo canceling** amount by inserting circuit loss into the line of a high frequency band and interrupting the high frequency component of a transmitting signal when howling is detected.

SOLUTION: This system is composed of an **acoustic echo removing device** 11 adopting a learning identifying method as adaptive algorithm, an analysis **filter** 12, a synthetic **filter** 13, a down-sampling circuit 14, an up-sampling circuit 15, a recoupling addition circuit 16, a howling detector 17, a **band limit** controller 18 and a circuit loss control circuit 19. When howling is detected by the howling detector 17, the successive update operation of a coefficient correction amount arithmetic circuit 7 on a high frequency band side is stopped, circuit loss is inserted into a pertinent high frequency band line and the high frequency component of a transmitting signal is interrupted. Thus, the probability of the generation of howling can be reduced.

15/5/2 (Item 2 from file: 347)  
DIALOG(R)File 347:JAPIO  
(c) 2003 JPO & JAPIO. All rts. reserv.

04604263 \*\*Image available\*\*  
DIGITAL **AUDIO** TRANSMITTING **DEVICE** AND RECEIVING DEVICE.

PUB. NO.: 06-276163 [JP 6276163 A]  
PUBLISHED: September 30, 1994 (19940930)  
INVENTOR(s): KATSUMATA TORU  
MATSUI JO  
APPLICANT(s): SONY CORP [000218] (A Japanese Company or Corporation), JP  
(Japan)  
APPL. NO.: 05-088116 [JP 9388116]  
FILED: March 23, 1993 (19930323)  
INTL CLASS: [5] H04B-014/04; H04N-007/13  
JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.6  
(COMMUNICATION -- Television)  
JOURNAL: Section: E, Section No. 1651, Vol. 18, No. 687, Pg. 86,  
December 26, 1994 (19941226)

#### ABSTRACT

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PURPOSE: To obtain digital audio transmitting and receiving devices whereby a circuit scale is made to be small by unifying the sampling frequency of an encoding system with different sampling frequencies.

CONSTITUTION: At the time of transmission, only the prescribed band of a signal supplied to a digital **low pass filter** 24 is supplied to a 1/2 thinning circuit 25 in a band converting part 2. Though the actual sampling frequency is reduced by the circuit 25, the **filter** 24 is provided before the circuit for satisfying a sampling theorem even at that time so that the band is limited. In the meantime, at the time of reception, a reception signal inputted to a signal separating part 31 with a digital line interface 30 is separated into a video signal and a sign code. The digital audio signal from a telephone quality voice decoder 32 is supplied to a sampling frequency converting part 10 and converted into the operation frequency of **echo canceller** 23. Thus, the audio signal is made to be the sampling frequency being the same as the audio signal from a high quality voice decoding part 36.

15/5/3 (Item 3 from file: 347)

DIALOG(R)File 347:JAPIO

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04064261 \*\*Image available\*\*

LOUD- **SPEAKER** SIMULTANEOUS SPEECH SYSTEM USING FREQUENCY DIVISION

PUB. NO.: 05-055961 [JP 5055961 A]

PUBLISHED: March 05, 1993 (19930305)

INVENTOR(s): ISHIKAWA KATSUKI

APPLICANT(s): AIPHONE CO LTD [324020] (A Japanese Company or Corporation),  
JP (Japan)

APPL. NO.: 03-215459 [JP 91215459]

FILED: August 27, 1991 (19910827)

INTL CLASS: [5] H04B-003/23; H04M-001/60

JAPIO CLASS: 44.2 (COMMUNICATION -- Transmission Systems); 44.4  
(COMMUNICATION -- Telephone)

JAPIO KEYWORD: R131 (INFORMATION PROCESSING -- Microcomputers &  
Microprocessors)

JOURNAL: Section: E, Section No. 1395, Vol. 17, No. 359, Pg. 63, July  
07, 1993 (19930707)

#### ABSTRACT

PURPOSE: To make the operation stable in double talking over the entire band of a voice signal by deciding the speech band respectively for the **echo canceller** system for reducing incoming and outgoing voice signals and for the system dividing the frequency depending on a cause of call occurrence for a high frequency so as to improve an **echo cancel** function.

CONSTITUTION: A low frequency component of a voice band at a caller side separated by a **low pass filter** 4 of a loudspeaking simultaneous speech equipment TE(sub 1) is converted into a digital signal and it is sent to a CPU 19. An **echo cancel** arithmetic operation program built in the CPU 19 references an outgoing voice signal from the **low pass filter** 11 to calculate a simulating echo. An incoming voice signal FS from which the simulating echo is subtracted is sent to a speech use D/A converter 6. On the other hand, a high frequency component in the incoming voice signal sent from an incoming caller side low frequency elimination **filter** 2 is eliminated in a way of an interdigital **filter** corresponding to the frequency of the outgoing voice signal FS of the caller **band pass filter** 9 so as to prevent howling of the high frequency component.

15/5/4 (Item 4 from file: 347)

DIALOG(R)File 347:JAPIO

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02252013     \*\*Image available\*\*  
EXHAUST SOUND REDUCING DEVICE FOR ENGINE

PUB. NO.:       62-168913 [JP 62168913 A]  
PUBLISHED:     July 25, 1987 (19870725)  
INVENTOR(s):   INOUE HIROSHI  
APPLICANT(s):  MAZDA MOTOR CORP [000313] (A Japanese Company or Corporation)  
                  , JP (Japan)  
APPL. NO.:     61-010507 [JP 8610507]  
FILED:         January 20, 1986 (19860120)  
INTL CLASS:    [4] F01N-001/00  
JAPIO CLASS:   21.2 (ENGINES & TURBINES, PRIME MOVERS -- Internal  
                  Combustion); 32.9 (POLLUTION CONTROL -- Other)  
JOURNAL:       Section: M, Section No. 657, Vol. 12, No. 4, Pg. 84, January  
                  08, 1988 (19880108)

ABSTRACT

PURPOSE: To **prevent** the acoustic **feedback** effect for reducing efficiently exhaust sounds, by entering a signal, which has a burning noise frequency corresponding to engine speed and a phase opposite to exhaust pulsation, into a pulsation generator.

CONSTITUTION: On an exhaust passage 1, a pressure sensor 2 is provided, and the signal from the sensor 2 is input in a sequential type probability control system 8 via a **band-pass filter** 3 and an A/D converter 4. Further, the signal from an engine speed sensor 5 is entered in a frequency converter 6, and in the converter 6 the frequency of engine speed pulse is converted to a basic frequency of burning sound. And, in the control system 8, a signal having a burning sound frequency and a phase opposite to exhaust pulsation is formed, and sent to a **speaker** 11. Thus, the acoustic **feedback** effect is **prevented**, and exhaust sounds can be reduced.

15/5/5        (Item 5 from file: 347)  
DIALOG(R)File 347:JAPIO  
(c) 2003 JPO & JAPIO. All rts. reserv.

01907524     \*\*Image available\*\*  
ECHO CONTROL SYSTEM

PUB. NO.:       61-121624 [JP 61121624 A]  
PUBLISHED:     June 09, 1986 (19860609)  
INVENTOR(s):   UMIGAMI SHIGEYUKI  
                  MURANO KAZUO  
                  KOSHIKAWA MASAMI  
APPLICANT(s):  FUJITSU LTD [000522] (A Japanese Company or Corporation), JP  
                  (Japan)  
APPL. NO.:     59-243808 [JP 84243808]  
FILED:         November 19, 1984 (19841119)  
INTL CLASS:    [4] H04B-003/20  
JAPIO CLASS:   44.2 (COMMUNICATION -- Transmission Systems)  
JOURNAL:       Section: E, Section No. 447, Vol. 10, No. 309, Pg. 65,  
                  October 21, 1986 (19861021)

ABSTRACT

PURPOSE: To realize an **echo canceller** with simplified circuit constitution by splitting a transmission system into a low-frequency signal component and a high-frequency signal component and using the **echo canceller** for the low frequency signal component to erase the echo component and using a comb-line **filter** with respect to the high frequency signal component to extract the frequency.

CONSTITUTION: After an incoming signal is converted into a digital signal,

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the result is separated into high and low frequency signals by a **band pass separation filter 12** of an incoming system. The separated high frequency component is subjected to spectral interleaving by a **comb-line filter 13** extracting the frequency component of a prescribed interval. The interleaving signal is added to the separated low frequency component at an adder **filter 14** to drive a **speaker 3**. The sound signal of a microphone 1 is separated into the high and low frequency signals by a **band pass filter 8** the same as that of the incoming system. The echo component is separated into the high and low frequencies, and the echo is blocked in terms of frequencies with respect to the high frequency component by the **comb-line filter 10**. The echo is eliminated with respect to the low frequency component by using a well-known **echo canceller 9**.

15/5/6 (Item 6 from file: 347)

DIALOG(R)File 347:JAPIO

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00413022

DRIVING CIRCUIT SYSTEM OF MULTIWAY **SPEAKERS**

PUB. NO.: 54-065022 [JP 54065022 A]

PUBLISHED: May 25, 1979 (19790525)

INVENTOR(s): TAKAHASHI NOBUAKI  
FUNASAKA EIICHI  
SHINOZAKI MASANOBU  
KAIZU YASUO

APPLICANT(s): VICTOR CO OF JAPAN LTD [000432] (A Japanese Company or Corporation), JP (Japan)

APPL. NO.: 52-131010 [JP 77131010]

FILED: November 01, 1977 (19771101)

INTL CLASS: [2] H04R-003/12; H03F-001/34

JAPIO CLASS: 42.5 (ELECTRONICS -- Equipment); 42.4 (ELECTRONICS -- Basic Circuits)

JOURNAL: Section: E, Section No. 125, Vol. 03, No. 86, Pg. 71, July 24, 1979 (19790724)

#### ABSTRACT

PURPOSE: To improve the damping characteristics of multiway **speakers** by adding and synthesizing plural signals having undergone frequency division and negative- feedback-operating this synthesized signal.

CONSTITUTION: The output signal of an amplifier 2 is applied to a tweeter 3 through a **high - pass filter**, to a squawker 4 through a **band - pass filter** and to a woofer 5 through a **low - pass filter**. The signals being applied to the tweeter 3, squawker 4 and woofer 5 after having been frequency-divided are respectively synthesized through resistances  $R(\text{sub } 1)$ ,  $R(\text{sub } 2)$ ,  $R(\text{sub } 3)$  and the synthesized signal is voltage-divided in resistances  $R(\text{sub } 4)$ ,  $R(\text{sub } 5)$ , after which it is applied to the inversion input terminal of the amplifier 2, whereby negative feedback is applied thereto. Then, even those up to the signals driving the **speakers 3, 4, 5** are included in the negative **feedback loop** and therefore, the degradation in the damping factors owing to the presence of equivalent series resistances in capacitors  $C(\text{sub } 1)$ ,  $C(\text{sub } 2)$ , coils  $L(\text{sub } 1)$ ,  $L(\text{sub } 2)$  may be eliminated.

15/5/7 (Item 1 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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014803722 \*\*Image available\*\*

WPI Acc No: 2002-624428/200267

Noise and echo canceling apparatus for use in hands free kit includes a high pass filter and a low pass filter sequentially

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**filter out non-audible frequency components of the user's voice signal from the microphone**

Patent Assignee: AEROTELECOM CO LTD (AERO-N); AERO TELECOM CO LTD (AERO-N)

Inventor: SEO G I; SUH G I

Number of Countries: 001 Number of Patents: 002

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
KR 2002012028	A	20020215	KR 200045447	A	20000805	200267 B
KR 335404	B	20020506	KR 200045447	A	20000805	200271

Priority Applications (No Type Date): KR 200045447 A 20000805

Patent Details:

Patent No	Kind	Lan Pg	Main IPC	Filing Notes
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KR 2002012028	A		1 H04R-003/02	
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KR 335404	B		H04R-003/02	Previous Publ. patent KR 2002012028
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Abstract (Basic): KR 2002012028 A

NOVELTY - The ground(7) is provided at a microphone(5) powered from the power supply(3) to ground a user's voice signal before transferring it to a noise canceler (6). The ground (9) connected to a **speaker** (8) to ground the other party's voice signal from the noise canceler(6). A **high pass filter** and a **low pass filter** sequentially **filter** out non-audible frequency components of the user's voice signal from the microphone (5). An analog switch turns on/off the output of the user's voice signal depending on the other party's voice signal.

DETAILED DESCRIPTION - In noise and **echo canceling** apparatus for a hands free, a power supply ground(4), a microphone ground(7) and a **speaker** ground(9) are separated from each other. The ground (7) is connected to a power supply (3) receiving power through a plug coupled to a cigarette lighter of a car to ground charge noise without effecting on other circuits.

USE - For hands-free apparatus of a GSM handheld phone.

ADVANTAGE - The noise and **echo canceling** is achieved by reducing effect of charge noise and grounds, amplifying audible frequencies only, and, if no signals at a **speaker**, outputting user's voice.

pp; 1 DwgNo 1/10

Title Terms: NOISE; ECHO; APPARATUS; HAND; FREE; KIT; HIGH; PASS; **FILTER**; LOW; PASS; **FILTER**; SEQUENCE; **FILTER**; NON; AUDIBLE; FREQUENCY; COMPONENT; USER; VOICE; SIGNAL; MICROPHONE

Derwent Class: P86; U25; W01

International Patent Class (Main): H04R-003/02

International Patent Class (Additional): G10K-011/00

File Segment: EPI; EngPI

15/5/8 (Item 2 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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012386052 \*\*Image available\*\*

WPI Acc No: 1999-192159/199917

XRPX Acc No: N99-140734

**Detection of direction of speech activity e.g. for hands-free speaker telephone**

Patent Assignee: NOKIA MOBILE PHONES LTD (OYNO )

Inventor: HAEKKINEN J; VALVE P; IIPPONEN P

Number of Countries: 026 Number of Patents: 003

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
EP 901267	A2	19990310	EP 98660083	A	19980826	199917 B
FI 9703596	A	19990305	FI 973596	A	19970904	199924
JP 11168791	A	19990622	JP 98249875	A	19980903	199935

Priority Applications (No Type Date): FI 973596 A 19970904

June 27, 2003

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes  
EP 901267 A2 E 14 H04M-009/08  
Designated States (Regional): AL AT BE CH CY DE DK ES FI FR GB GR IE IT  
LI LT LU LV MC MK NL PT RO SE SI  
JP 11168791 A 42 H04R-003/00  
FI 9703596 A H04B-000/00

Abstract (Basic): EP 901267 A

NOVELTY - Microphones, preferably four or more, in a microphone vector (2) receive a voice signal and each produces a single signal. This is passed through **band - pass filters** (14) to direction angle estimating to units (15, 17), that store the assumed direction of arrival of the voice. This is then compared to the assumed direction in a detection unit (18), to identify a match of the assumed and estimated directions of arrival of a voice signal.

DETAILED DESCRIPTION - An INDEPENDENT CLAIM is included for a detection device of voice sources.

USE - Detecting source of voice using receiving microphone e.g. in full duplex **speaker** phones that are prone to echo.

ADVANTAGE - Achieves **echo - canceling** and suppresses information of about message-end speech activity

DESCRIPTION OF DRAWING(S) - The drawing is a block diagram of detector according to present invention.

Microphone vector 2

**Band - pass filter** 14

Estimation and recording units 15, 17

Direction detection unit 18

Dwg.2/9

Title Terms: DETECT; DIRECTION; SPEECH; ACTIVE; HAND; FREE; **SPEAKER** ;  
TELEPHONE

Derwent Class: P86; W01; W04

International Patent Class (Main): H04B-000/00; H04M-009/08; H04R-003/00

International Patent Class (Additional): G01H-003/00; G10L-009/00;

H04M-001/60

File Segment: EPI; EngPI

15/5/9 (Item 3 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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011468484 \*\*Image available\*\*

WPI Acc No: 1997-446391/199741

XRPX Acc No: N97-372012

**Adaptive type noise removal appts for vehicle telephone - has adaptive filters , which vary characteristics of each signal path of transmission signal to which presumed echo is added, adaptively based on adder's output**

Patent Assignee: NEC CORP (NIDE )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 9205388	A	19970805	JP 9611574	A	19960126	199741 B

Priority Applications (No Type Date): JP 9611574 A 19960126

Patent Details:

Patent No Kind Lan Pg Main IPC Filing Notes  
JP 9205388 A 6 H04B-003/23

Abstract (Basic): JP 9205388 A

The appts consists of an acoustic **echo canceller** (12) between a radio (6) and a **speaker** (11). The output transmission signals from a microphone (1) is passed to the radio. The acoustic **echo canceller** forms a presumed **echo** based on the answering signal. Presumed echo is

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added to the transmission signal through an adder (4) and the acoustic echo component is removed from the transmission signal.

A first and second adaptive **high pass filters** (14,15) are provided at front stage of adder and adaptive **echo canceller** respectively. The variation of the characteristics of each signal path of the transmission signal is adaptively done by the adaptive **high pass filter**, based on the output of the adder.

ADVANTAGE - Improves S/N of transmission signal. Reduces acoustic echoes. Improves interactive quality and transmission articulation.

Dwg.1/5

Title Terms: ADAPT; TYPE; NOISE; REMOVE; APPARATUS; VEHICLE; TELEPHONE; ADAPT; **FILTER**; VARY; CHARACTERISTIC; SIGNAL; PATH; TRANSMISSION; SIGNAL; ECHO; ADD; ADAPT; BASED; ADDER; OUTPUT

Derwent Class: U22; W01

International Patent Class (Main): H04B-003/23

International Patent Class (Additional): H03H-017/00; H03H-021/00; H04M-001/60

File Segment: EPI

15/5/10 (Item 4 from file: 350)

DIALOG(R) File 350:Derwent WPIX

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010250988 \*\*Image available\*\*

WPI Acc No: 1995-152243/199520

XRPX Acc No: N95-119680

**Noise reduction device for air conditioner - incorporates pair of speakers and microphone in ventilation duct to impart symmetry to acoustic characteristic of ventilation duct about vertical axis**

Patent Assignee: MATSUSHITA DENKI SANGYO KK (MATU )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
JP 7077994	A	19950320	JP 93222702	A	19930908	199520 B

Priority Applications (No Type Date): JP 93222702 A 19930908

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
JP 7077994	A		13	G10K-011/178	

Abstract (Basic): JP 7077994 A

The noise reduction device has several FIR **filters**. Two microphones (1a,1b) and two **speakers** (5a,5b) are located in the ventilation duct (9) attached to the blower (8). The first microphone (1a) detects the acoustic noise emitted by the air blower. The LMS calculator (4) executes the signal processing of detected acoustic noise. The adaptive **filter** (2) executes the adaptive control of the calculator output.

The two **speakers** (5a,5b) generates acoustic output from the adaptive **filter** electrical output. The second digital **filter** (3b) processes the signal outputted by the adaptive **filter** and feeds the result to the LMS calculator (4). The second acoustic noise detector i.e. microphone is installed in the ventilation duct at far end near the **speaker**. The acoustic noise detected by the second microphone is fed to the LMS calculator (4). The **speakers** (5a,5b) are arranged so that the cones face each other, ensuring axial alignment with direction of airflow.

ADVANTAGE - Improves noise reduction effect and **low pass** reproduction capability with low cost **speakers**. Provides uniform and stable attenuation effect. **Prevents feedback loop** constituted by output signal of second digital **filter** and adaptive **filter** reproduced by calculator from **speaker** is detected by first microphone. Provides stable noise control. Delivers good acoustic damping. Regulates flow of air between first and second detection

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points.

Dwg.1/9

Title Terms: NOISE; REDUCE; DEVICE; AIR; CONDITION; INCORPORATE; PAIR;  
**SPEAKER** ; MICROPHONE; VENTILATION; DUCT; IMPART; SYMMETRICAL; ACOUSTIC;  
CHARACTERISTIC; VENTILATION; DUCT; VERTICAL; AXIS

Derwent Class: P86; Q74; U22; W04; X27

International Patent Class (Main): G10K-011/178

International Patent Class (Additional): F24F-013/02; H03H-017/02;  
H03H-021/00

File Segment: EPI; EngPI

15/5/11 (Item 5 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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009041337 \*\*Image available\*\*

WPI Acc No: 1992-168696/199221

XRPX Acc No: N92-127146

**RF signal transceiver for remote access to vehicles - uses RF oscillator stage with feedback loop incorporating saw delay line and control switching between transmitting and receiving modes**

Patent Assignee: DELCO ELECTRONICS CORP (DELC-N)

Inventor: ANDERSON F J

Number of Countries: 006 Number of Patents: 009

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week	
EP 486087	A2	19920520	EP 91202848	A	19911101	199221	B
AU 9187030	A	19920611	AU 9187030	A	19911106	199232	
US 5146613	A	19920908	US 90614488	A	19901116	199239	
JP 4269014	A	19920925	JP 91300357	A	19911115	199245	
EP 486087	A3	19921125	EP 91202848	A	19911101	199343	
NZ 240600	A	19940427	NZ 240600	A	19911114	199420	
EP 486087	B1	19950906	EP 91202848	A	19911101	199540	
DE 69112774	E	19951012	DE 612774	A	19911101	199546	
			EP 91202848	A	19911101		
KR 9507493	B1	19950711	KR 9120448	A	19911116	199715	

Priority Applications (No Type Date): US 90614488 A 19901116

Cited Patents: No-SR.Pub; 1.Jnl.Ref; FR 2165740; US 4786903

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
EP 486087	A2	E	6	H04B-001/40	
US 5146613	A		7	H04B-001/44	
JP 4269014	A		5	H04B-001/44	
EP 486087	B1	E	7	H04B-001/40	
DE 69112774	E			H04B-001/40	Based on patent EP 486087
AU 9187030	A			H04B-001/40	
EP 486087	A3			H04B-001/40	
NZ 240600	A			H04B-001/40	
KR 9507493	B1			H04B-001/44	

Abstract (Basic): EP 486087 A

The transceiver has an RF oscillator (20) which includes a feedback circuit (22,24) comprising a surface **acoustic** wave **device** (24) coupling an input and an output of the RF oscillator, and adapted to produce RF oscillations. A controller (14) switches the appts. between transmitting and receiving modes. A transmitter (38,40,12) is coupled to the output of the RF oscillator during the transmitting mode to transmit the RF oscillations produced.

An input (12,16,18) is adapted during the receiving mode to couple a modulated RF signal to the input of the RF oscillator. A **low pass filter** (34) is coupled to the output of the RF oscillator, and adapted to **filter** the modulated RF signal.

ADVANTAGE - Small size, low power consumption, and good high temp.

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stability.

Dwg.1-3/6

Title Terms: RF; SIGNAL; TRANSCEIVER; REMOTE; ACCESS; VEHICLE; RF;  
OSCILLATOR; STAGE; FEEDBACK; LOOP; INCORPORATE; SAW; DELAY; LINE; CONTROL  
; SWITCH; TRANSMIT; RECEIVE; MODE

Derwent Class: W05; X22

International Patent Class (Main): H04B-001/40; H04B-001/44

International Patent Class (Additional): H04B-001/30

File Segment: EPI

15/5/12 (Item 6 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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008423815

WPI Acc No: 1990-310816/199041

XRFX Acc No: N90-238370

**Extended bandwidth virtual earth noise controller - has filter between  
speaker and microphone moved nearer noise source, so transport lead  
component reduces loop gain requirements**

Patent Assignee: ANONYMOUS (ANON )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
RD 317102	A	19900910				199041 B

Priority Applications (No Type Date): RD 90317102 A 19900820

Abstract (Basic): RD 317102 A

The virtual earth active noise controller consists of a microphone for measuring acoustic noise generated by a source, an inverting amplifier having the microphone connected to its input, and a **speaker** connected to the output of the inverting amplifier. The noise signal measured by the input microphone is inverted by the amplifier, and corresponding acoustic waves are produced by the **speaker** to cancel the noise waves. The input microphone is moved a slight distance away from the **speaker** and toward the source of noise. A **filtering** network is included between the input microphone and **speaker**.

The measured signal produced by the input microphone then consists of a transport lead component due to the measured noise, and a **loop feedback** component due to the waves produced by the **speaker**. The transport lead component reduces the loop gain requirements for optimum noise reduction and stability. The total system noise attenuation is significantly greater than the loop gain and is attributable to the transport lead component. The **filtering** network is designed to provide the proper amplification and phase compensation, in accord with the transfer functions of the other system components, such as the microphone and **speaker**, to increase the loop gain at the **bandpass limits**.

ADVANTAGE - Operable noise canceling bandwidth of noise controller is significantly increased.

Dwg.0/0

Title Terms: EXTEND; BANDWIDTH; VIRTUAL; EARTH; NOISE; CONTROL; **FILTER** ;  
**SPEAKER** ; MICROPHONE; MOVE; NEARBY; NOISE; SOURCE; SO; TRANSPORT; LEAD;  
COMPONENT; REDUCE; LOOP; GAIN; REQUIRE

Derwent Class: P86; W04

International Patent Class (Additional): G10K-000/00

File Segment: EPI; EngPI

15/5/13 (Item 7 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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004181455

WPI Acc No: 1985-008335/198502

XRPX Acc No: N85-005840

**Sound attenuation system using active control techniques - has controller incorporating signal processor feeding signal from sound detector to generator using negative feedback**

Patent Assignee: NAT RES DEV CORP (NATR )

Inventor: SWINBANKS M A

Number of Countries: 003 Number of Patents: 004

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
GB 2142091	A	19850109	GB 8415387	A	19840615	198502 B
JP 60020700	A	19850201	JP 84128374	A	19840621	198511
US 4589133	A	19860513	US 84620751	A	19840614	198622
GB 2142091	B	19870325				198712

Priority Applications (No Type Date): GB 8317086 A 19830623; GB 8415387 A 19840615

Patent Details:

Patent No	Kind	Lan	Pg	Main IPC	Filing Notes
GB 2142091	A		8		

Abstract (Basic): GB 2142091 A

In an active sound control system allowance is made in a relatively uncomplicated circuit for acoustic coupling between a sound generating system for generating a cancelling sound wave and a detector for sensing a sound wave to be cancelled. Unwanted sound from a source is detected by a microphone and cancelled by sound from a **speaker**.

The microphone is connected to the **speaker** by way of a fixed gain amplifier which has a feedback processing system. The system is such that its transfer function takes account of acoustic feedback between the **speaker** and the microphone in deriving, with the amplifier 9, a signal to drive the **speaker**.

USE - For cancelling unwanted sound.

3/6

Title Terms: SOUND; ATTENUATE; SYSTEM; ACTIVE; CONTROL; TECHNIQUE; CONTROL; INCORPORATE; SIGNAL; PROCESSOR; FEED; SIGNAL; SOUND; DETECT; GENERATOR; NEGATIVE; FEEDBACK

Derwent Class: P86; Q51; W04

International Patent Class (Additional): F01N-001/06; G10K-011/16; H04R-003/04

File Segment: EPI; EngPI

15/5/14 (Item 8 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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001767091

WPI Acc No: 1977-L3606Y/197751

**Band pass filter with surface acoustic wave devices - uses LC circuit to suppress triple transit echoes in surface wave receiver output**

Patent Assignee: ROCKWELL INT CORP (ROCW )

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 4063202	A	19771213				197751 B

Priority Applications (No Type Date): US 76683608 A 19760505

Abstract (Basic): US 4063202 A

At least one of several transducers on a piezoelectric substrate is externally short-circuited by a series resonance circuit for suppression of triple transit echoes. The resonance circuit is additionally connected to an amplifier via a high input or output

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impedance.

In multi-transducer systems any two transducers are **prevented** from communicating via **echo** signals by such resonance circuits. In **filter** cascades several resonance circuits may be included in the interstage coupling circuit or between the input and output amplifiers on the one hand and the cascade on the other hand.

Title Terms: BAND; PASS; **FILTER** ; SURFACE; ACOUSTIC; WAVE; DEVICE; CIRCUIT ; SUPPRESS; TRIPLE; TRANSIT; ECHO; SURFACE; WAVE; RECEIVE; OUTPUT

Derwent Class: U25

International Patent Class (Additional): H03H-009/26; H03H-013/00

File Segment: EPI

15/5/15 (Item 9 from file: 350)

DIALOG(R)File 350:Derwent WPIX

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001459950

WPI Acc No: 1976-C2843X/197610

**Coin discriminating appts using coin vibration - has selective oscillator running at natural coin frequency and one-cycle selector**

Patent Assignee: MITANI SHOJI KK (MITA-N)

Number of Countries: 001 Number of Patents: 001

Patent Family:

Patent No	Kind	Date	Applicat No	Kind	Date	Week
US 3939953	A	19760224				197610 B

Priority Applications (No Type Date): JP 7368678 A 19730620; JP 7368676 A 19730620; JP 7368677 A 19730620

Abstract (Basic): US 3939953 A

The coin discriminating apparatus contains an oscillator circuit having a **feedback loop**. A mechanical **filter** includes a discriminated coin, a **speaker** to vibrate the coin and a sensor to pick up the vibration of the coin. A one cycle selector takes out one period of the vibration frequency generated at the oscillator circuit. Means are provided for quantizing the output signal of the one cycle selector by clock pulses. Counter means includes a scale of -1000 counter and a decoder for counting the number of clock pulses. A bistable circuit has its state reversed on receipt of an output produced at the decoder when contents of the counter run up to the **lower limit** or the upper limit of a predetermined tolerance.

Title Terms: COIN; APPARATUS; COIN; VIBRATION; SELECT; OSCILLATOR; RUN; NATURAL; COIN; FREQUENCY; ONE; CYCLE; SELECT

Derwent Class: T05

International Patent Class (Additional): G07F-003/02

File Segment: EPI

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File 348:EUROPEAN PATENTS 1978-2003/Jun W04

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File 349:PCT FULLTEXT 1979-2002/UB=20030626,UT=20030619

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Set	Items	Description
S1	39503	(AUDIO? OR AMPLIFICATION? OR ACOUSTIC? OR HEARING) (2N) (AID? OR DEVICE? OR APPARATUS? OR APPTS) OR SPEAKER?
S2	311902	FILTER?
S3	67065	(BAND? OR HIGH? OR LOW?) () (PASS OR STOP? OR LIMIT?) OR PAS- SBAND?
S4	18783	(FEEDBACK? OR ECHO?) (3N) (LOOP? OR CANCEL? OR PREVENT?)
S5	222	S1(3N)S4
S6	3	S5(S)S2(S)S3
S7	1227	S2(5N)S4
S8	9	S7(S)S1(S)S3
S9	8	S8 NOT S6
S10	7166	S2(3N)ADAPT?
S11	5	S10(S)S1(S)S3(S)S4
S12	2	S11 NOT (S9 OR S6)
S13	39	S1(S)S2(S)S3(S)S4
S14	5	S13 AND IC=H04R-025/00
S15	1	S14 NOT (S12 OR S9 OR S6)

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6/5,K/1 (Item 1 from file: 348)  
DIALOG(R)File 348:EUROPEAN PATENTS  
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00782736

**HEARING AID  
HORHILFSGERAT  
PROTHESE AUDITIVE**

**PATENT ASSIGNEE:**

TOPHOLM & WESTERMANN APS, (374270), Ny Vestergaardsvej 25, DK-3500.  
Vaerloese, (DK), (applicant designated states: AT;CH;DE;DK;IT;LI;NL)

**INVENTOR:**

ANDERSEN, Henning, Haugaard, Adalsvej 40, DK-2970 Horsholm, (DK)

**LEGAL REPRESENTATIVE:**

Bohmer, Hans Erich, Dipl.-Ing. (2312), Keplerstrasse 23, 71134 Aidlingen,  
(DE)

PATENT (CC, No, Kind, Date): EP 793897 A1 970910 (Basic)  
EP 793897 B1 980513  
WO 9617493 960606

APPLICATION (CC, No, Date): EP 95921771 950529; WO 95EP2033 950529

PRIORITY (CC, No, Date): DE 4441996 941126

DESIGNATED STATES: AT; CH; DE; DK; IT; LI; NL

INTERNATIONAL PATENT CLASS: H04R-025/00;

**NOTE:**

No A-document published by EPO

**LEGAL STATUS (Type, Pub Date, Kind, Text):**

Application: 960911 A International application (Art. 158(1))

Application: 970910 A1 Published application (A1with Search Report  
;A2without Search Report)

Examination: 970910 A1 Date of filing of request for examination:  
970502

Examination: 971112 A1 Date of despatch of first examination report:  
971001

Grant: 980513 B1 Granted patent

Oppn None: 990506 B1 No opposition filed

LANGUAGE (Publication,Procedural,Application): German; German; German

**FULLTEXT AVAILABILITY:**

Available Text	Language	Update	Word Count
CLAIMS B	(English)	9820	324
CLAIMS B	(German)	9820	271
CLAIMS B	(French)	9820	326
SPEC B	(German)	9820	1722
Total word count - document A			0
Total word count - document B			2643
Total word count - documents A + B			2643

...CLAIMS 3), essentially consists of a subtraction stage (5) with a positive and negative input, a **low - pass filter** (6) and a comparator circuit (7) with holding network controlled by a clock pulse generator...

...the subtraction stage to the output of the comparator stage (7) by way of a **feedback loop**.

3. **Hearing aid** in accordance with claim 1, characterized in that the clock frequency of the clock pulses...

6/5,K/2 (Item 2 from file: 348)  
DIALOG(R)File 348:EUROPEAN PATENTS  
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00271449

**Superregenerative detector.  
Pendelruckkopplungs-Detektor.**

June 27, 2003

**Detecteur a super-reaction.**

**PATENT ASSIGNEE:**

R.F. MONOLITHICS, INC., (327971), 4441 Sigma Road, Dallas Texas 75244,  
(US), (applicant designated states: DE;FR;GB;IT;NL)

**INVENTOR:**

ASH, Darrell L., 1707 Cartwright Dr. Sachse, Texas 75040, (US)

**LEGAL REPRESENTATIVE:**

Rackham, Stephen Neil et al (35061), GILL JENNINGS & EVERY, Broadgate  
House, 7 Eldon Street, London EC2M 7LH, (GB)

PATENT (CC, No, Kind, Date): EP 271190 A2 880615 (Basic)  
EP 271190 A3 890531  
EP 271190 B1 940302

APPLICATION (CC, No, Date): EP 87308927 871008;

PRIORITY (CC, No, Date): US 939527 861208

DESIGNATED STATES: DE; FR; GB; IT; NL

INTERNATIONAL PATENT CLASS: H03D-011/04;

CITED PATENTS (EP A): EP 184508 A; US 3405364 A; US 3119065 A; US 4143324 A  
; FR 2209255 A

**CITED REFERENCES (EP A):**

IEEE TRANSACTIONS ON CONSUMER ELECTRONICS, vol. CE-33, no. 3, August  
1987, pages 395-404, New York, US; D.L. ASH: " A low cost  
superregenerative saw stabilized receiver"  
IDEM;

**ABSTRACT EP 271190 A2**

A superregenerative detector utilizing a single transistor and having a  
surface acoustic wave device in the feed back loop coupling the output to  
the input to cause oscillation wherein the surface acoustic wave device  
is a low loss delay line formed as a single phase unidirectional  
transducer on a quartz substrate.

ABSTRACT WORD COUNT: 55

**LEGAL STATUS (Type, Pub Date, Kind, Text):**

Application: 880615 A2 Published application (A1with Search Report  
;A2without Search Report)  
Search Report: 890531 A3 Separate publication of the European or  
International search report  
Examination: 891220 A2 Date of filing of request for examination:  
891023  
Examination: 911016 A2 Date of despatch of first examination report:  
910902  
Grant: 940302 B1 Granted patent  
Oppn None: 950222 B1 No opposition filed

LANGUAGE (Publication,Procedural,Application): English; English; English

**FULLTEXT AVAILABILITY:**

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	147
CLAIMS B	(German)	EPBBF1	142
CLAIMS B	(French)	EPBBF1	174
SPEC B	(English)	EPBBF1	2492
Total word count - document A			0
Total word count - document B			2955
Total word count - documents A + B			2955

...SPECIFICATION switching the RF oscillator between an oscillating and a  
non-oscillating condition; characterised by a **surface acoustic wave**  
delay line device in the **feedback loop**, the **device** being a **single**  
phase **unidirectional** transducer formed on a piezoelectric substrate  
with electrodes a quarter of a wavelength wide, and...

...there are means for coupling a modulated RF signal to the oscillator  
input; and a **low pass filter** means coupled to the output to recover  
the modulation signal.

In the accompanying drawings:-

FIG...

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6/5,K/3 (Item 1 from file: 349)  
DIALOG(R)File 349:PCT FULLTEXT  
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00556232 \*\*Image available\*\*

**BAND-LIMITED ADAPTIVE FEEDBACK CANCELLER FOR HEARING AIDS**  
**DISPOSITIF ADAPTATIF DE SUPPRESSION DE L'EFFET LARSEN A BANDE LIMITEE**  
**DESTINE AUX PROTHESES AUDITIVES**

Patent Applicant/Assignee:

HOUSE EAR INSTITUTE,

Inventor(s):

GAO Shawn,

SOLI Sigfrid,

CHI Hsiang-Feng,

Patent and Priority Information (Country, Number, Date):

Patent: WO 200019605 A2 20000406 (WO 0019605)

Application: WO 99US22757 19990930 (PCT/WO US9922757)

Priority Application: US 98102557 19980930

Designated States: AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ CZ DE

DE DK DK DM EE EE ES FI FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP

KR KZ LC LK LR LS LT LU LV MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG

SI SK SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW GH GM KE LS MW SD SL SZ

TZ UG ZW AM AZ BY KG KZ MD RU TJ TM AT BE CH CY DE DK ES FI FR GB GR IE

IT LU MC NL PT SE BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

Main International Patent Class: H04R-025/00

International Patent Class: H04R-003/02

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 9255

**English Abstract**

An improved method for adaptively cancelling acoustic feedback in hearing aids and other audio amplification devices. Feedback cancellation is limited to a frequency band that encompasses all unstable frequencies. By limiting the bandwidth of the feedback cancellation signal, the distortion due to the adaptive filter is minimized and limited only to the unstable feedback regions. A relatively simple signal processing algorithm is used to produce highly effective results with minimal signal distortion.

**French Abstract**

L'invention concerne un procede ameliore pour supprimer de maniere adaptative l'effet Larsen dans les protheses auditives et dans d'autres dispositifs audio amplifies. La suppression de l'effet Larsen est limitee a la bande de frequences qui englobe toutes les frequences instables. En limitant la bande de frequences du signal d'annulation de l'effet Larsen, on arrive a reduire au minimum la distorsion provoquee par le filtre adaptatif, qui est limitee uniquement aux zones instables de l'effet Larsen. On utilise un algorithme relativement simple de traitement des signaux pour obtenir des resultats probants, et ce avec une distorsion minimale des signaux.

Fulltext Availability:

Claims

**Claim**

I A feedback canceller for an audio amplification device comprising:

an adaptive filter ;

means for combining an output of the adaptive filter with an input of the audio amplification device;

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a first **band limiting filter** having an input coupled to an output of the audio amplification device and an output coupled to an input of the adaptive **filter**, wherein the first **band limiting filter** has a **passband** limited to a frequency band containing unstable frequencies.  
2 The device of claim 1 wherein...

...the output of the audio amplification device to the input thereof.

25 A method for **cancelling feedback** in an **audio amplification device** comprising the steps of  
applying an output of the audio amplification device to a first **band limiting filter** having a **passband** limited to a frequency band containing unstable frequencies.  
applying an output of the first **band limiting filter** to an adaptive **filter** ;  
combining an output of the adaptive **filter** with an input of the audio amplification device.

26 The method of claim 25 wherein...path from the output of the audio amplification device to the input thereof

48 A **feedback canceller** for an **audio amplification device** comprising:  
means for creating a first delay having an input coupled to an audio output of a hearing aid circuit and an output, &  
a first **band limiting filter** having an input coupled to the output of the first delay means and an output;  
an adaptive **filter** having an input coupled to the output of the first **band limiting filter** and an output;  
means for creating a second delay having an input coupled to a...

...of the second delay means, an inverting input coupled to the output of the adaptive **filter** and an output coupled to the input of the hearing aid processing module;  
a second **band limiting filter** having an input coupled to the input of the second delay means and an output...

...second summing node having a non-inverting input coupled to the output of the second **band limiting filter**, an inverting input coupled to the output of the adaptive **filter** and an output;  
means for selecting a **filter** coefficient having a first input coupled to the output of the first **band limiting filter**, a second input coupled to the output of the second summing node and an output for supplying the **filter** coefficient to the adaptive **filter** ;  
wherein the first and second **band limiting filters** have **passbands** limited to a frequency band containing unstable frequencies.

49 A **feedback canceller circuit** for an **audio amplification device** comprising:  
means for creating a delay having an input coupled to an audio output of a hearing aid circuit and an output;  
a first **band limiting filter** having an input coupled to the output of the delay means and an output;  
an adaptive **filter** having an input coupled to the output of the first

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**band limiting filter** and an output;  
a summing node having a non-inverting input coupled to a  
conditioned...

...of a hearing aid microphone, an inverting input coupled to the output of  
the adaptive **filter** and an output coupled to the input of  
the hearing aid processing module;  
a second **band limiting filter** having an input coupled to the output  
of the summing node and an output;  
a third **band limiting filter** having an input coupled to the output  
of  
the first **band limiting filter** and an output;  
means for selecting a **filter** coefficient having a first input coupled  
to the output of the second **band limiting filter** and a second input  
coupled to the output of the third **band limiting filter** and an  
output for supplying  
the **filter** coefficient to the adaptive **filter** ;  
wherein the first, second and third **band limiting** filters have  
**passbands** limited to a frequency band containing unstable frequencies.

50 The device of claim 49 wherein...

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9/5,K/1 (Item 1 from file: 348)  
DIALOG(R)File 348:EUROPEAN PATENTS  
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01534470

**ECHO PROCESSING APPARATUS**  
**VORRICHTUNG ZUR ECHOVERARBEITUNG**  
**APPAREIL DE TRAITEMENT D'ECHOS**

**PATENT ASSIGNEE:**

MITSUBISHI DENKI KABUSHIKI KAISHA, (208589), 2-3, Marunouchi 2-chome,  
Chiyoda-ku, Tokyo 100-8310, (JP), (Applicant designated States: all)

**INVENTOR:**

TAKAHASHI, Shinya, Mitsubishi Denki K.K., 2-3, Marunouchi 2-chome,  
Chiyoda-ku, Tokyo 100-8310, (JP)  
MATSUOKA, Bunkei, Mitsubishi Denki K.K., 2-3, Marunouchi 2-chome,  
Chiyoda-ku, Tokyo 100-8310, (JP)  
KAJIYAMA, Ikuo, Mitsubishi Denki Engineering K.K., 6-2, Otemachi 2-chome,  
Chiyoda-ku, Toyko 100-0004, (JP)

**LEGAL REPRESENTATIVE:**

Pfenning, Meinig & Partner (100961), Mozartstrasse 17, 80336 Munchen,  
(DE)

PATENT (CC, No, Kind, Date): EP 1300963 A1 030409 (Basic)  
WO 2002095975 021128

APPLICATION (CC, No, Date): EP 2002771733 020520; WO 2002JP4860 020520

PRIORITY (CC, No, Date): JP 2001152888 010522

DESIGNATED STATES: AT; BE; CH; CY; DE; DK; ES; FI; FR; GB; GR; IE; IT; LI;  
LU; MC; NL; PT; SE; TR

EXTENDED DESIGNATED STATES: AL; LT; LV; MK; RO; SI

INTERNATIONAL PATENT CLASS: H04B-003/23; H04M-001/60; H04R-003/02;  
G10L-021/02

**ABSTRACT EP 1300963 A1**

An **echo canceller** has a **high - pass filter** 11, a **speaker** 3, an A/D converter 6, and an **echo canceller** 7. The **high - pass filter** 11 suppresses a low-frequency component in a received signal. The **speaker** 3 outputs an acoustic sound of a low-frequency suppressed received signal passed through the **high - pass filter** 11. The A/D converter 6 converts an acousticecho from a microphone 4 into a transmission signal of a digital form. The echo canceller 7 generates a pseudo echo signal based on the low-frequency suppressed received signal passed through the **high - pass filter** 11, and eliminates the acoustic echo to be inputted to the microphone 4 from the **speaker** 3 by subtracting the pseudo echo signal from the digital signal converted by the A/D converter 6.

ABSTRACT WORD COUNT: 122

**NOTE:**

Figure number on first page: 1

**LEGAL STATUS (Type, Pub Date, Kind, Text):**

Application: 030122 A1 International application. (Art. 158(1))

Application: 030122 A1 International application entering European phase

Application: 030409 A1 Published application with search report

Examination: 030409 A1 Date of request for examination: 20030110

LANGUAGE (Publication,Procedural,Application): English; English; Japanese

**FULLTEXT AVAILABILITY:**

Available Text	Language	Update	Word Count
CLAIMS A	(English)	200315	692
SPEC A	(English)	200315	7925
Total word count - document A			8617
Total word count - document B			0
Total word count - documents A + B			8617

**...ABSTRACT A1**

An **echo canceller** has a **high - pass filter** 11, a **speaker** 3,

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an A/D converter 6, and an **echo canceller** 7. The **high - pass filter** 11 suppresses a low-frequency component in a received signal..The **speaker** 3 outputs an acoustic sound of a low-frequency suppressed received signal passed through the **high - pass filter** 11. The A/D converter 6 converts an acousticecho fromamicrophone 4 into a transmission signal...

...a pseudo echo signal based on the low-frequency suppressed received signal passed through the **high - pass filter** 11, and eliminates the acoustic echo to be inputted to the microphone 4 from the **speaker** 3 by subtracting the pseudo echo signal from the digital signal converted by the A...

...SPECIFICATION processor having a high-pass filter, aD/Aconverter, a speaker, amicrophone, anA/Dconverter, and an **echo canceller** . The **high - pass filter** suppress a low-frequency component in a received signal in digital form. The D/A converter converts the low-frequency component passed through the **high - pass filter** to a sound signal. The **speaker** outputs an acoustic based on the sound signal. The microphone has a possibility to input an acoustic echo outputted from the **speaker** . The A/D converter converts the sound signal outputted from the microphone to a digital...

...a pseudo echo signal based on a low-frequency suppressed received signal obtained through the **high - pass filter** and generates a transmission signal by subtracting the pseudo echo signal from the digital...

...suppressed received signal to be inputted to the echo canceller is suppressed together by the **high - pass filter**, it is possible to prevent a deterioration of a calculation accuracy of an adaptive **filter** coefficient in the **echo canceller** , to set the difference between the pseudo echo signal and the echo signal to a...

...to another aspect of the present invention, there is provided an echo processor having a **high - pass filter**, a D/A converter, a speaker, a microphone, an A/D converter, an echo canceller, and a double-talk detector. The **high - pass filter** suppress a low-frequency component in a received signal in digital form. The D/A converter converts the low-frequency component passed through the **high - pass filter** to a sound signal. The speaker outputs an acoustic based on the sound signal ...

...a pseudo echo signal based on a low-frequency suppressed received signal obtained through the **high - pass filter** and generates a transmission signal by subtracting the pseudo echo signal from the digital...

...on the low-frequency component, and controls to halt and start the update of a **filter** coefficient of the **echo canceller** .  
It is thereby possible to reduce a non-linear distortion outputted from the speaker. Further...

...suppressed received signal to be inputted to the echo canceller is suppressed together by the **high - pass filter**, it is possible to prevent a deterioration of a calculation accuracy of an adaptive **filter** coefficient in the **echo canceller** , to set the difference between the pseudo echo signal and the echo signal to a...

9/5,K/2 (Item 2 from file: 348)  
DIALOG(R)File 348:EUROPEAN PATENTS  
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00734692

Hands-free communication apparatus with echo canceler  
Freisprechanlage mit Echokompensator

June 27, 2003

**Appareil de communication moins libres avec annuleur d'echo**

**PATENT ASSIGNEE:**

    MITSUBISHI DENKI KABUSHIKI KAISHA, (208580), 2-3, Marunouchi 2-chome  
    Chiyoda-ku, Tokyo 100, (JP), (applicant designated states: DE;FR;GB)

**INVENTOR:**

    Higuchi, Koji, c/o Mitsubishi Denki K.K., Tsushinki Seisakusho, 1-1  
    Tsukaguchi-honmachi, 8-chome, Amagasaki-shi, Hyogo 661, (JP)  
    Shiono, Takashi, c/o Mitsubishi Denki K.K., Tsushinki Seisakusho, 1-1  
    Tsukaguchi-honmachi, 8-chome, Amagasaki-shi, Hyogo 661, (JP)

**LEGAL REPRESENTATIVE:**

    Sajda, Wolf E., Dipl.-Phys. et al (9956), MEISSNER, BOLTE & PARTNER  
    Postfach 86 06 24, D-81633 Munchen, (DE)

PATENT (CC, No, Kind, Date): EP 692903 A1 960117 (Basic)

APPLICATION (CC, No, Date): EP 95110937 950712;

PRIORITY (CC, No, Date): JP 94161522 940713

DESIGNATED STATES: DE; FR; GB

INTERNATIONAL PATENT CLASS: H04M-009/08;

ABSTRACT EP 692903 A1

    A hands-free speaking device is provided which is capable of removing ambient noise at the near side from the speech transmitting signal.

    The hands-free speaking device comprises a speaker (51) for reproducing a received speech signal, a microphone (52) for outputting a speech transmitting signal, an echo canceler (55) for removing an echo (62) transmitted from the speaker (51) to the microphone (52), and a voice detector (11) comprising a noise level monitoring circuit (2) for monitoring the level of the ambient noise (63) input to the microphone (52), a voice level monitoring circuit (3) for monitoring the level of a voice signal (61) input to the microphone (52), and a comparator (4) for generating a signal for increasing the attenuation amount of an attenuator (10) to attenuate the transmitting speech signal when the noise level is higher than the voice level. (see image in original document)

ABSTRACT WORD COUNT: 170

**LEGAL STATUS (Type, Pub Date, Kind, Text):**

    Examination: 020327 A1 Date of dispatch of the first examination report: 20020206

    Application: 960117 A1 Published application (A1with Search Report ;A2without Search Report)

    Examination: 960327 A1 Date of filing of request for examination: 960124

LANGUAGE (Publication,Procedural,Application): English; English; English

**FULLTEXT AVAILABILITY:**

Available Text	Language	Update	Word Count
CLAIMS A	(English)	EPAB96	537
SPEC A	(English)	EPAB96	3695
Total word count - document A			4232
Total word count - document B			0
Total word count - documents A + B			4232

...SPECIFICATION 54, an acoustic echo canceler 55, and a 2-line/4-line converter 57.

    The **high - pass** filter 12 attenuates the low-frequency component in the speech transmitting signal generated from the speech transmitting signal microphone 52. The high-frequency component alone is passed through the **filter** and output to the acoustic **echo canceler** 55. The **low - pass filter** 13 attenuates the high-frequency component in the output of the acoustic echo canceler 55...

...to the attenuator 10. The attenuator 10, the voice detector 11, the received speech signal **speaker** 51, the speech transmitting signal microphone 52, the speech transmitting side output terminal 53, the...

June 27, 2003

DIALOG(R)File 348:EUROPEAN PATENTS  
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00252199

**Programmable sound reproducing system.**

**Programmierbares Schallwiedergabesystem.**

**Systeme de reproduction sonore programmable.**

PATENT ASSIGNEE:

AUDIMAX CORPORATION, (1522870), c/o Energy Transportation Group, Inc.  
1185 Avenue of the Americas, New York, New York 10036, (US), (applicant  
designated states: AT;BE;CH;DE;ES;FR;GB;GR;IT;LI;LU;NL;SE)

INVENTOR:

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Dugot, Richard S., 590 West End Avenue, New York New York 10024, (US)  
Kopper, Kenneth W., 746 Oak Avenue, River Edge New Jersey 07661, (US)

LEGAL REPRESENTATIVE:

Rackham, Stephen Neil (35061), GILL JENNINGS & EVERY, Broadgate House, 7  
Eldon Street, London EC2M 7LH, (GB)

PATENT (CC, No, Kind, Date): EP 250679 A2 880107 (Basic)  
EP 250679 A3 890419  
EP 250679 B1 930707

APPLICATION (CC, No, Date): EP 86307717 861007;

PRIORITY (CC, No, Date): US 879214 860626

DESIGNATED STATES: AT; BE; CH; DE; ES; FR; GB; GR; IT; LI; LU; NL; SE

INTERNATIONAL PATENT CLASS: H04R-025/00; H03G-005/16; H04R-003/02;

CITED PATENTS (EP A): US 4188667 A; EP 71845 A; GB 1582821 A; US 4131760 A;  
NL 8500595 A

ABSTRACT EP 250679 A2

In a hearing aid system, selected optimum parameter values are programmed into an electronically erasable, programmable read-only memory (EEPROM) (84) which supplies coefficients to a programmable filter (64) and amplitude limiter (67) in the hearing aid so as to cause the hearing aid to adjust automatically to the optimum set of parameter values for the speech level, room reverberation, and type of background noise then obtaining. The programmable filter may be a digital equivalent of a tapped delay line in which each delayed sample is multiplied by a weighting coefficient, and the sum of the weighted samples generates a desired electro-acoustic characteristic; or a tapped analog delay line in which the sum of the weighted outputs of the taps generates the desired characteristics.

Acoustical feedback is reduced by an electrical feedback path in the hearing aid which is matched in both amplitude and phase to the acoustic feedback path, the two feedback signals being subtracted so as to cancel each other. Alternatively, a single filter in the forward path may be used for this purpose with a transmission characteristic equivalent to that of the programmable filter in the forward path plus the electrical feedback path. Also, the relative speech-noise content in the signals from the hearing aid microphone is sensed and binary words are generated and supplied to the programmable filter for selecting from memory a set of delay line tap coefficients that are effective to impart to the filter the appropriate frequency response for the specific environmental noise condition being detected.

ABSTRACT WORD COUNT: 256

LEGAL STATUS (Type, Pub Date, Kind, Text):

Lapse: 20000126 B1 Date of lapse of European Patent in a  
contracting state (Country, date): GR  
19930707, LU 19931031,  
Application: 880107 A2 Published application (A1with Search Report  
;A2without Search Report)  
Change: 880831 A2 Representative (change)  
Search Report: 890419 A3 Separate publication of the European or  
International search report  
Examination: 891129 A2 Date of filing of request for examination:

June 27, 2003

891004  
Examination: 910807 A2 Date of despatch of first examination report:  
910620  
Change: 920902 A2 Representative (change)  
\*Assignee: 920902 A2 Applicant (transfer of rights) (change):  
AUDIMAX CORPORATION (1522870) c/o Energy  
Transportation Group, Inc. 1185 Avenue of the  
Americas New York, New York 10036 (US)  
(applicant designated states:  
AT;BE;CH;DE;ES;FR;GB;GR;IT;LI;LU;NL;SE)  
Grant: 930707 B1 Granted patent  
Oppn None: 940629 B1 No opposition filed  
Lapse: 991229 B1 Date of lapse of European Patent in a  
contracting state (Country, date): LU  
19931031,

LANGUAGE (Publication,Procedural,Application): English; English; English

FULLTEXT AVAILABILITY:

Available Text	Language	Update	Word Count
CLAIMS B	(English)	EPBBF1	1366
CLAIMS B	(German)	EPBBF1	1487
CLAIMS B	(French)	EPBBF1	1773
SPEC B	(English)	EPBBF1	6717

Total word count - document A 0

Total word count - document B 11343

Total word count - documents A + B 11343

...SPECIFICATION aid, switching back and forth two sets of electroacoustic characteristics at will by means of the switches 123A and 123B, choosing the characteristic which is more intelligible or preferable in some way. Paired comparisons made...

...88 is connected to the switch 37, the counter 74 and the RAM 71.

The hearing aid is now in its normal operating mode and speech detected by the microphone 57 is...

...gain control circuit 58 and transmitted through the amplifier 60, the filter 63 and the low pass filter 63a into the programmable filter circuitry.

A so-called "bucket brigade" analog delay time...

9/5,K/4 (Item 1 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

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00776439 \*\*Image available\*\*

DC OFFSET CALIBRATION FOR A DIGITAL SWITCHING AMPLIFIER

CALIBRAGE DE DECALAGE EN CONTINU POUR AMPLIFICATEUR DE COMMUTATION  
NUMERIQUE

Patent Applicant/Assignee:

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95054, US, US (Residence), US (Nationality)

Inventor(s):

MIAO Guoqing, 529 W. Field Avenue, Roselle Park, NJ 07204, US  
DELANO Cary, 1531 Fairway Green Circle, San Jose, CA 95131, US

Legal Representative:

VILLENEUVE Joseph M, Beyer Weaver & Thomas, LLP, P.O. Box 130, Mountain  
View, CA 94042-0130, US

Patent and Priority Information (Country, Number, Date):

Patent: WO 200110012 A1 20010208 (WO 0110012)

Application: WO 2000US20197 20000725 (PCT/WO US0020197)

Priority Application: US 99146416 19990729; US 2000624503 20000724

Designated States: AE AG AL AM AT AU AZ BA BB BG BR BY BZ CA CH CN CR CU CZ  
DE DK DM DZ EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ  
LC LK LR LS LT LU LV MA MD ME MK MN MW MX MZ NO NZ PL PT RO RU SD SE SG  
SI SK SL TJ TM TR TT TZ UA UG UZ VN YU ZA ZW

June 27, 2003

(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE  
(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG  
(AP) GH GM KE LS MW MZ SD SL SZ TZ UG ZW  
(EA) AM AZ BY KG KZ MD RU TJ TM

Main International Patent Class: H03F-001/02

International Patent Class: H03F-003/217; G01R-019/00

Publication Language: English

Filing Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 4594

#### English Abstract

An offset voltage calibration circuit for use with a digital switching amplifier (400). The calibration circuit includes an analog-to-digital converter (406) for converting at least one DC offset voltage associated with the digital switching amplifier (400) to digital offset data. A memory (408) stores the digital offset data. Control circuitry (402) controls the analog-to-digital converter (406). A digital-to-analog converter (404) coupled to the memory (408) receives the digital offset data and generates an offset compensation voltage for applying to an input port of the digital switching amplifier which thereby cancels at least a portion of the at least one DC offset voltage.

#### French Abstract

L'invention concerne un circuit de calibrage de tension de decalage a utiliser avec un amplificateur (400) de commutation numerique. Ledit circuit de calibrage comprend un convertisseur analogique-numerique (406) permettant de convertir au moins une tension de decalage en continu associee a l'amplificateur (400) de commutation numerique en donnees de decalage numerique. Une memoire (408) stocke les donnees de decalage numerique. Des circuits de commande (402) commandent le convertisseur analogique-numerique (406). Un convertisseur numerique-analogique (404) relie a la memoire (408) recoit les donnees de decalage numerique et genere une tension de compensation de decalage a appliquer a une borne d'entree de l'amplificateur de commutation numerique qui annule ainsi au moins une partie d'une tension de decalage en continu au minimum.

#### Legal Status (Type, Date, Text)

Publication 20010208 A1 With international search report.

Examination 20010614 Request for preliminary examination prior to end of 19th month from priority date

Fulltext Availability:

Detailed Description

#### Detailed Description

... amplifier I 00 and converted to a one-bit signal by a noise-shaping oversampled **feedback loop** which includes **loop filter** 102, comparator 104, break-before-make generator 106, power stage driver 108, and power stage...

...The one-bit signal drives the power stage I IO which, in turn, drives a **low pass** filter comprising inductor 1 1 2 and capacitor 1 1 4 which recovers the audio signal with which **speaker** 1 1 6 is driven.

Any DC offset inherent in amplifier I 00 is amplified...

9/5,K/5 (Item 2 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

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00749056 \*\*Image available\*\*

GATEWAY WITH VOICE

June 27, 2003

**PASSERELLE VOCALE**

**Patent Applicant/Assignee:**

BROADCOM CORPORATION, 16215 Alton Parkway, Irvine, CA 92618-3616, US, US  
(Residence), US (Nationality), (For all designated states except: US)

**Patent Applicant/Inventor:**

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, CA, CA (Residence), CA (Nationality), (Designated only for: US)  
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1J3, CA, CA (Residence), CA (Nationality), (Designated only for: US)  
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HARTMAN David, 16215 Alton Parkway, Irvine, CA 92618-3616, US, US  
(Residence), US (Nationality), (Designated only for: US)

**Legal Representative:**

GELFOUND Craig A (agent), Christie, Parker & Hale, LLP, P.O. Box 7068,  
Pasadena, CA 91109-7068, US,

**Patent and Priority Information (Country, Number, Date):**

Patent: WO 200062501 A2-A3 20001019 (WO 0062501)  
Application: WO 2000US10149 20000413 (PCT/WO US0010149)  
Priority Application: US 99129134 19990413; US 99136685 19990528; US  
99154903 19990920; US 99156266 19990927; US 99157470 19991001; US  
99160124 19991018; US 99161152 19991022; US 99162315 19991028; US  
99163169 19991102; US 99163170 19991102; US 99163600 19991104; US  
99164379 19991109; US 99164689 19991110; US 99164690 19991110; US  
99166289 19991118; US 99454219 19991209; US 99171203 19991215; US  
99171169 19991216; US 99171180 19991216; US 99171184 19991216; US  
2000178258 20000125; US 2000493458 20000128; US 2000522185 20000309

**Designated States:** AE AL AM AT AU AZ BA BB BG BR BY CA CH CN CR CU CZ DE DK

DM EE ES FI GB GD GE GH GM HR HU ID IL IN IS JP KE KG KP KR KZ LC LK LR  
LS LT LU LV MA MD MG MK MN MW MX NO NZ PL PT RO RU SD SE SG SI SK SL TJ  
TM TR TT TZ UA UG US UZ VN YU ZA ZW

(EP) AT BE CH CY DE DK ES FI FR GB GR IE IT LU MC NL PT SE

(OA) BF BJ CF CG CI CM GA GN GW ML MR NE SN TD TG

(AP) GH GM KE LS MW SD SL SZ TZ UG ZW

(EA) AM AZ BY KG KZ MD RU TJ TM

Main International Patent Class: H04L-012/28

International Patent Class: H04L-012/66; H04L-007/02; H04B-003/23

Publication Language: English

Filing Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 80268

June 27, 2003

English Abstract

In one aspect of the present invention, a network gateway is configured to facilitate on line and off line bi-directional communication between a number of near end data and telephony devices with far end data termination devices via a hybrid fiber coaxial network and a cable modem termination system. The described network gateway combines a QAM receiver, a transmitter, a DOCSIS MAC, a CPU, a voice and audio processor, an Ethernet MAC, and a USB controller to provide high performance and robust operation.

French Abstract

Selon un aspect de la presente invention, une passerelle de reseau est concue pour faciliter la communication bidirectionnelle en-ligne et hors-ligne entre, d'une part, une pluralite de dispositifs de telephonie et de traitement de donnees d'extremite rapprochee et, d'autre part, des dispositifs terminaux de traitement de donnees d'extremite eloignee, par l'intermediaire d'un reseau a systeme de transmission hybride fibre et coaxial et d'un systeme de terminaison a modem cable. La passerelle de reseau de cette invention combine un recepteur QAM, un emetteur, un MAC DOCSIS, une unite centrale, un processeur de donnees vocales et sonores, un MAC Ethernet et un controleur USB dans le but d'assurer de hautes performances et un fonctionnement robuste.

Legal Status (Type, Date, Text)

Publication 20001019 A2 Without international search report and to be republished upon receipt of that report.  
Examination 20010412 Request for preliminary examination prior to end of 19th month from priority date  
Search Rpt 20020103 Late publication of international search report  
Republication 20020103 A3 With international search report.

Fulltext Availability:  
Detailed Description

Detailed Description

... embodiment of the present invention;  
FIG. 18A is a block diagram of a single pole **low pass** filter used to smooth or average the differences between sampling rates in accordance with a...  
...typically associated with conventional echo cancellers and utilizes the delay associated with a decimator and **high pass** filter to provide a look ahead capability so that filter I 0 adaptation may be...  
...the present invention;  
I 0 FIG. 31 is a block diagram of a method for **canceling echo** returns in accordance with a

9/5,K/6 (Item 3 from file: 349)  
DIALOG(R) File 349:PCT FULLTEXT  
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00514343 \*\*Image available\*\*

METHOD AND APPARATUS FOR MONITORING, CONTROLLING, AND CONFIGURING REMOTE COMMUNICATION DEVICES

PROCEDE ET APPAREIL DE CONTROLE, DE COMMANDE ET DE CONFIGURATION DE DISPOSITIFS DE COMMUNICATION DISTANTS

Patent Applicant/Assignee:

CONEXANT SYSTEMS INC,

Inventor(s):

COLLIN Zeev,  
TAMIR Tal,

June 27, 2003

Patent and Priority Information (Country, Number, Date):

Patent: WO 9945695 A1 19990910  
Application: WO 99US4841 19990304 (PCT/WO US9904841)  
Priority Application: US 9876784 19980304; US 98154643 19980917; US  
98193304 19981117

Designated States: AL AM AU AZ BA BB BG BR BY CA CN CU CZ EE GE GH GM HR HU  
ID IL IS JP KE KG KP KR KZ LC LK LR LS LT LV MD MG MK MN MW MX NO NZ PL  
RO RU SD SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW AT BE CH CY DE DK ES  
FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04M-011/06

International Patent Class: H04L-012/26; H04L-012/24

Publication Language: English

Fulltext Availability:

Detailed Description

Claims

Fulltext Word Count: 8571

English Abstract

A communication system for monitoring and/or controlling communication parameters of a remote communication device. The communication system monitors a communication channel that is created between the remote communication device and controls the communication device by adjusting internal settings of the communication device that represent communication parameters. The communication device is communicatively coupled to a communication channel to carry out ongoing communications between the communication device and the communication channel. Further, a software module is associated with the communication device, and the software module accesses the internal settings of the communication device from a remote location via the communication channel and performs diagnostics such as monitoring, controlling, and configuring the communication device using the internal settings of the communication device.

French Abstract

Système de communication destiné à contrôler et/ou commander les paramètres de communication d'un dispositif de communication distant. Le système de communication contrôle une voie de communication créée entre le dispositif de communication distant et il commande le dispositif de communication en ajustant les réglages internes du dispositif de communication représentant les paramètres de communication. Le dispositif de communication est couplé de manière communicative à une voie de communication afin de permettre des communications permanentes entre le dispositif de communication et la voie de communication. De plus, un module logiciel est associé au dispositif de communication, le module logiciel accède au réglage interne du dispositif de communication depuis un point distant par la voie de communication et il exécute un diagnostic tel que le contrôle, la commande et la configuration du dispositif de communication au moyen des réglages internes de ce dernier.

Fulltext Availability:

Claims

Claim

... 0 indicates

u-law

RKCFG -ENCODING-LAW V90-RK-CODES, BOOL

(TRUE=A-Law) None

\*\*\*\*\* **SpeakerPhone** Constants

Hardware Delay

RKCFG -EC-DELAY SPKP-RK-CODES, //@SPKP

MODULE,INT No of

Samples...

...RKMON DATA RES ECHO GET

RKCTL-DATA-RES-ECHO-REQUEST=v34-RK-CODESI// None

None

June 27, 2003

**SpeakerPhone Constants**

**Speakerphone** Mode (FD, HD, HS)

RKCTL-SPKP-MODE SPKP-RK-CODES, SPKPMode

None

Output Mute

RKCTL IO MUTE,

@SPKP PROBE,BOOL - Yes/No None

// **Echo Cancellers**

RKCTL **FILTER** -LENGTH, @SPKP-MODULE,INT - No

of Taps) None

RKCTL EC-OPERATE, (SPKP

MODULE,BOOL

Yes...bit)

PCM Pad

RKMON-PAD-DETECTED,

None DWORD PAD 0=NORMAL 3=3dBPad 6=6dBPad

// **High Pass** filter enabled

RKMON-HIGHPASS-FILTER-ENABLED

None BOOL Yes/No

**SpeakerPhone Constants**

**Speakerphone** Mode (FD, HD, HS)

RKMON-SPKP-MODE SPKP-RK-CODES, None

so SPKPMode

State

RKMON...

...Yes/No

RKMON-SATURATION,

SPKP PROBE BOOL - Yes/No

RKMON-DC-LEVEL,

SPKP PROBE FLOAT

**Echo Cancellers**

RKMON- **FILTER** -LENGTH,

SPKP MODULE INT No of Taps

RKMON-EC-OPERATEI

SPKP-MODULE BOOL Yes/No...

9/5,K/7 (Item 4 from file: 349)

DIALOG(R)File 349:PCT FULLTEXT

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00514342 \*\*Image available\*\*

**METHOD AND APPARATUS FOR MONITORING, CONTROLLING, AND CONFIGURING LOCAL COMMUNICATION DEVICES**

**PROCEDE ET APPAREIL DE SURVEILLANCE, REGLAGE ET CONFIGURATION DE DISPOSITIFS LOCAUX DE TRANSMISSION**

Patent Applicant/Assignee:

CONEXANT SYSTEMS INC,

Inventor(s):

COLLIN Zeev,

TAMIR Tal,

Patent and Priority Information (Country, Number, Date):

Patent: WO 9945694 A1 19990910

Application: WO 99US4690 19990304 (PCT/WO US9904690)

Priority Application: US 9876784 19980304; US 98154643 19980917; US

98192627 19981116

Designated States: AL AM AU AZ BA BB BG BR BY CA CN CU CZ EE GE GH GM HR HU

ID IL IS JP KE KG KP KR KZ LC LK LR LS LT LV MD MG MK MN MW MX NO NZ PL

RO RU SD SG SI SK SL TJ TM TR TT UA UG UZ VN YU ZW AT BE CH CY DE DK ES

FI FR GB GR IE IT LU MC NL PT SE

Main International Patent Class: H04M-011/06

International Patent Class: H04L-012/26; H04L-012/24

Publication Language: English

Fulltext Availability:

June 27, 2003

Detailed Description  
Claims

Fulltext Word Count: 8979

English Abstract

A communication system for monitoring and/or controlling communication parameters of a communication device. The communication system monitors a communication channel that is created when the communication device connects to a network, controls the communication device as it operates on the network, and configures the communication device. The communication device is commonly a modem and is communicatively coupled to the network to carry out ongoing communications between the modem and the network through the communication channel. Further, a software module is associated with the modem, and the software module accesses the internal settings of the modem via the communication channel (if necessary) and performs operations such as monitoring, controlling, and configuring the modem (or other communication device) using the internal settings of the modem.

French Abstract

L'invention porte sur un systeme de telecommunications surveillant et/ou regulant les parametres d'un systeme de telecommunications. Ledit systeme surveille un canal de transmission cree au moment ou le dispositif de telecommunications se raccorde a un reseau, le commande alors qu'il opere sur le reseau, et le configure. Le dispositif de telecommunications est habituellement un modem raccorde au reseau et acheminant les communications en cours entre le modem et le reseau et transitant par le canal de transmission. En outre, un logiciel associe au modem et ayant acces aux reglages interieurs du modem (si necessaire par l'intermediaire du canal de transmission), execute des operations telles que la surveillance, le reglage, ou la configuration du modem (ou d'autres dispositifs de transmission) en utilisant les reglages interieurs du modem.

Fulltext Availability:  
Claims

Claim

... 0 indicates  
u-law  
RKCFG-ENCODING-LAW V90-RK-CODES, BOOL  
(TRUE=A-Law) None  
\*\*\*\*\* **SpeakerPhone** Constants  
Hardware Delay  
RKCFG-EC- DELAY SPKP-RK-CODES, //(SPKP  
MODULE,INT No of  
Samplesi...  
...RKMON DATA RES ECHO GET  
RKCTL-DATA-RES-ECHO-REQUEST=v34-RK-CODES, // None  
None  
    **SpeakerPhone** Constants  
    **Speakerphone** Mode (FD, HD, HS)  
RKCTL-SPKP-MODE SPKP-RK-CODES, SPKPMode  
None  
Output Mute  
RKCTL-TO-MUTE,  
@SPKP-PROBEIBOOL - Yes/No@ None  
// **Echo** **Cancellers**  
RKCTL- **FILTER** -LENGTH, @SPKP-MODULE,INT - No  
of Taps1 None  
RKCTL-EC-OPERATE, @SPKP-MODULE,BOOL  
Yes...bit)  
PCM Pad  
RKMON- PAD-DETECTED,

June 27, 2003

None DWORD PAD 0=NORMAL 3=3dBPad 6=6dBPad  
// **High Pass** filter enabled  
RKMON-HIGHPASS-FTLTER-ENABLED  
None BOOL Yes/No  
**SpeakerPhone** Constants  
**Speakerphone** Mode (FD, HD, HS)  
RKMON-SPKP-MODE SPKP-RK-CODES, None  
SPKPMode  
State  
RKMON-STATE...

...Yes/No  
RKMON-SATURATION,  
SPKP-PROBE BOOL - Yes/No  
RKMON-DC-LEVELI  
SPKP-PROBE FLOAT  
// **Echo Cancellers**  
RKMON- **FILTER** -LENGTHI  
SPKP MODULE INT No of Taps  
RKMON-EC-OPERATE,  
SPKP MODULE BOOL Yes/No...

9/5,K/8 (Item 5 from file: 349)  
DIALOG(R)File 349:PCT FULLTEXT  
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00191306

**HEARING AID**

**PROTHESE AUDITIVE**

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Detailed Description  
Claims

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**English Abstract**

Programmable hybrid hearing aid with digital signal processing comprising a main section (1) which can be inserted in the meatus (6). The main section (1) comprises an open connection between the ear opening and an inner portion of the meatus (6), providing an acoustic transmission channel with low-pass characteristic and resonant amplification. The main section further comprises an electroacoustic transmission channel based on digital signal processing and a signal processor (DSP) and with possibility for suppressing a possible acoustic signal feedback through the acoustic transmission channel. A variant of the hearing aid is provided with a microphone (M1) and the feedback signal is suppressed by digital filtering. Another variant of the hearing

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aid employs two microphones (M1, M2), and the feedback signal may then be suppressed by phasing out before the digital signal processing, while the digital signal processing also comprises cancellation of the feedback signal in case of high gain. A number of response functions are stored in a memory (RAM2) in a control unit and is freely chosen by the user in regard of adaption to hearing function and acoustic environment. All the electronics of the electroacoustic channel in the hearing aid is implemented as a monolithic integrated circuit (3) in CMOS technology.

#### French Abstract

Prothese auditive hybride programmable avec traitement des signaux numeriques. Elle comprend une partie principale (1) a introduire dans le conduit auditif (6). Celle-ci (1) comporte une liaison ouverte entre l'orifice de l'oreille et la partie interne du conduit auditif (6), constituant ainsi un canal de transmission acoustique a effet de passe-bas et d'amplification par resonance. Cette partie principale comprend en outre un canal de transmission electroacoustique base sur le traitement de signaux numeriques, ainsi qu'une unite de traitement des signaux. Il est en outre possible de supprimer une eventuelle reaction parasite acoustique grace au canal de transmission acoustique. Une autre version de cette prothese auditive prevoit un microphone (M1), le signal de reaction etant supprime par filtrage numerique. Dans une troisieme version, on a deux microphones (M1, M2), et le signal de reaction peut alors etre supprime par elimination progressive de phase avant le traitement du signal numerique, qui entraine aussi l'elimination du signal de retour en cas de gain eleve. Un certain nombre de fonctions de reaction sont memorisees (RAM2) dans une unite de commande. L'utilisateur peut librement les selectionner selon leur degre d'adaptation a la fonction auditive et a l'environnement acoustique. Toutes les pieces electroniques du canal electro-acoustique de ladite prothese sont realisees sous forme de circuit integre monolithe (3) par technique CMOS.

#### Fulltext Availability:

Claims

#### Claim

achieved with a **hearing aid** which is characterized by the features presented by the characteristic part of claim 5.  
A method for detection and signal processing in a **hearing aid** principally of the type presented in claim 5. is characterized by the features presented by the characteristic part of claim 13,  
Further features and advantages of the **hearing aid** in accordance with the invention are presented in the appended independent claims 2, 4 and...

...with the attached drawings.

Fig. 1a is a block diagram showing the principles of a **hearing aid** in accordance with the invention.

Fig. 1b is a schematic representation of an electrical equivalent connection for the acoustic channel in fig. 1a.

Fig. 2 is a variant of the **hearing aid** in accordance with the invention.

Fig. 3 is a further variant of the **hearing aid** in accordance with the invention.

Fig. 4a is a schematic block diagram for a **hearing aid** in accordance with the invention, where one microphone is used.

Fig. 4b shows the **hearing aid** in fig. 4a with a cancellation **filter** inserted in a **feedback loop**.

Fig. 4c shows the **hearing aid** in fig. 4a with a cancellation filter inserted in the signal's forward path.

Fig. 4d shows the **hearing aid** in fig. 4a with a power amplifier in the output stage.

Fig. 5a shows a **hearing aid** in accordance with the invention, where two microphones are used,

Fig. 5b shows a digital signal processor used with the **hearing aid** in fig. 5a.

Fig. 6a is three examples of response curves for strong, moderate and weak hearing impairment respectively, in addition to the sound pressure response of a meatus without **hearing aid**.

Fig. 6b is an example of the response curve for an envelope signal and a...

...in

the digital signal processor in fig. 5b,

The principles of the design of a **hearing aid** in accordance with the invention are illustrated schematically in fig. 1a.

The **hearing aid** comprises an electroacoustic channel consisting of an analog input section, a digital signal processor and...

...analog output section together with an acoustic transmission channel which simultaneously constitutes both an acoustic **low pass** filter and a potential acoustic feedback path, An external sound field is detected by a to the detector via its acoustic channel, The method of construction of the **hearing aid** causes a section of the inner meatus near the tympanum also to constitute an active component of the **hearing aid** by acting as a resonator.

The acoustic channel will be discussed in more detail in connection with the equivalence diagram in fig. 1b..

Fig. 2 shows a variant of the **hearing aid** in accordance with the invention. This variant comprises a main section with an acoustic transmission...

...a distance from the first microphone M1. The electronic components which form part of the **hearing aid** are provided in a first secondary section 2a which here is positioned in the concha...

...this secondary section 2a it may be appropriate to provide a battery 4 for the **hearing aid**. Another not shown secondary section constitutes a case for the **hearing aid**, on an inner end of the main section 1 is provided a miniaturised sound generator SG which faces the tympanum and converts the amplified electrical signal in the **hearing aid** to an **acoustic** signal which is intercepted by the tympanum, In order to have room inside a person...

...must preferably have a diameter which is less than approximately 4,5 mm. In the **hearing aid** in accordance with the present invention an electrodynamic sound generator of the type described in fig. 3 the **hearing aid** in accordance with the invention is shown in a variant with two microphones M1 and...

...to the main section 1 in fig. 2. All the electronics as well as the **hearing aid**'s battery 4 are provided in the main section 1, so that a secondary section provided in or beside the concha has been dispensed with, The **hearing aid**'s main section 1 has rather been connected with a not shown secondary section 2 in the form of a case in which the main section is kept when the **hearing aid** is not in use and which may also comprise possible electronic and electrical auxiliary devices...

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- ...possibly plugs and switches and is arranged so that it is used for charging the **hearing aid** 's battery 4 when . . . the main section I is in the case. The main section...
- ...charging.  
The electrical and electronic components used for signal processing in a variant of the **hearing aid** in accordance with the invention with one microphone M1, will now be described in more...
- ...fig. 4a, All of these components can be provided in a suitable manner in the **hearing aid** 's main section 1 or possibly in a first secondary section 2a, The microphone M1...
- ...frequency of, e.g., 8 kHz. This will therefore be the upper limit of the **hearing aid** 's frequency response, The microphone M1 may be, e.g., a cardioid micropohone which gives...front of the inputs of a sound generator SG.'In order to eliminate any acoustic **feedback** a **cancellation filter** 35 is used which in fig, 4b is shown inserted in a feedback loop between...
- ...amplifier 15 are all connected to a battery 4 which is preferably provided in the **hearing aid** 's main section 1.  
Fig. 5a shows the electronic components for signal processing in a **hearing aid** in accordance with the invention which uses two microphones M1, M2. In the figure the...
- ...the input to the first channel  
CH1 and a second channel CH2 respectively in the **hearing aid** 's analog section, Each channel CH1, CH2 thus comprises a series connection of an impedance...

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00616101

**METHOD FOR FAIL-SAFE OPERATION IN A SPEAKER PHONE SYSTEM**  
**AUSFALLGESICHERTES BETRIEBSVERFAHREN IN EINEM LAUTFERNSPRECHSYSTEM**  
**PROCEDE DE FONCTIONNEMENT A SECURITE INTEGREE DANS UN SYSTEME TELEPHONIQUE**  
**A HAUT PARLEUR**

**PATENT ASSIGNEE:**

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Assignee: 020807 A1 Transfer of rights to new applicant: Polycom,  
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SPEC B	(English)	200238	3671
Total word count - document A			0
Total word count - document B			4122
Total word count - documents A + B			4122

...SPECIFICATION Reference Manual" by AT&T, Oct. 1989.

Speaker signal 45 from telephone lines 44 is low - pass filtered and  
digitized at 8 kilohertz by A/D converter 47. The digitized speaker  
signal 51 is combined with Line Echo Canceler (LEC) signal 49 in

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summer 53 to produce summer output 55. As in AEC 26, LEC 40 uses an **adaptive filter** to model the impulse response of the sidetones coming from the near-end signals via microphone 12. The replica of the sidetones is then eliminated from the **speaker** signals 51 to prevent the near-end talkers from hearing their own voices coming back. Similar to the **adaptive filter** in AEC 26, a conventional normalized least mean square algorithm is used. Summer output 55 is fed back to LEC 40 to measure the effectiveness of the **echo cancellation**.

Following **echo cancellation** on the receive side, summer output 55 connects to receive attenuator 57, which functions in...

12/5,K/2 (Item 1 from file: 349)  
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00253802

**METHOD FOR FAIL-SAFE OPERATION IN A SPEAKER PHONE SYSTEM**

**PROCEDE DE FONCTIONNEMENT A SECURITE INTEGREE DANS UN SYSTEME TELEPHONIQUE A HAUT PARLEUR**

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Patent and Priority Information (Country, Number, Date):

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Detailed Description

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**English Abstract**

A method is disclosed for maintaining loop stability between a transmit and a receive path of a speaker phone which is capable of operating in either a full-duplex or a half-duplex mode. Loop stability is maintained and normal operation is achieved by controlling transmit and receive path attenuators. The method first determines whether feedforward or feedback signals levels should be used (76). Based on this determination, the signal and noise parameters along various points in the transmit and receive path are evaluated (78, 80). Based on the parameter values (82), the transmit and receive attenuators are adjusted to maintain loop stability and to operate the speaker phone in the proper state (84, 75).

**French Abstract**

L'invention se rapporte a un procede permettant de maintenir la stabilite de boucle entre une voie de transmission et une voie de reception d'un poste telephonique a haut-parleur pouvant fonctionner soit en mode duplex integral soit en mode semi-duplex. La stabilite de boucle est maintenue, et un fonctionnement normal est obtenu par la regulation d'affaiblisseurs de voies de transmission et de reception. Le procede consiste a determiner tout d'abord si des niveaux de signaux a reaction vers l'avant ou de retour devraient etre utilises (76). En fonction de cette determination, les parametres de signaux et de bruits au niveau de differents points situes le long des voies de transmission et de reception sont evalues (78, 80). En fonction de ces valeurs parametriques (82), les affaiblisseurs de transmission et de reception sont ajustes afin de maintenir la stabilite de boucle et de faire fonctionner le poste

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00498434

HEARING AID.

HORGERAT.

PROTHESE AUDITIVE.

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...SPECIFICATION the optimum adaptation of the auditory signal to any  
hearing residue and simultaneously optimize the **hearing aid** 's  
response function, **hearing aids** have been developed wherein the  
signal processing is performed digitally. The response function is  
adapted through filtering of the digital signal by means of appropriate  
**filter** coefficients, thus permitting the frequency response to some

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- extent to simulate the response function of a person with normal **hearing**. If the **aids** of the digital type are designed as so-called all-in-the-ear aids, the...
- ...it will be an advantage to have several response curves, in order to adapt the **hearing aid**'s **amplification** as a function of the frequency to a variety of acoustic environments. It is obvious...
- ...the range from approximately 1 up to approximately 4 kHz.
- Another well-known problem with **hearing aids**, whether they are digital or analog, is acoustic feedback between sound generator and microphone. Even though the **hearing aid** is positioned so that it closes the meatus and thus also prevents utilization of any hearing residue, this does not **prevent feedback** at high amplification, since the sound from the sound generator can be conducted back to the microphone either via the material of the **hearing aid** or via tissue and bone matter in the vicinity of the meatus. It will therefore be desirable to **cancel** such a **feedback** signal, e.g. in connection with the digital signal processing in the **hearing aid**. As has already been mentioned it is also desirable to utilize any hearing residue at...
- ...at least partially open, preferably so that it creates an acoustic transmission channel with a **low - pass** characteristic between the ear opening and the tympanum. If a channel of this kind is to be used with a **hearing aid** of the all-in-the-ear type, this makes great demands on the miniaturization of the **hearing aid**. Moreover, the problem of acoustic feedback will be further accentuated and will need to be...the ear opening to the inner meatus.
- This acoustic transmission channel ATC functions as a **low - pass filter** whose characteristics in reality depend on the volume of the channel and the volume of...
- ...converted to an analog output signal  $s(\text{sub}(r))$  which is smoothed in the reconstruction **filter** 14. The output signal from the reconstruction **filter** 14 is conveyed to the input terminals of the sound generator SG whose acoustic output...
- ...case of high amplifications, e.g. over 55 dB, it will therefore be necessary to **cancel** this **feedback** signal, which is done preferably by means of a cancellation **filter** 35 in the digital signal processor DSP. The cancellation is performed in a purely digital manner in the cancellation **filter** 35 which can be provided in various ways in the digital signal processor, e.g. in a **feedback loop** between the output from the equalizer 34 and the input of the compressor 33 as...
- ...for detection and signal processing in accordance with the invention involving the use of a **hearing aid** with two microphones will now be described in more detail with reference to figs. 5a...
- ...dB, in case the amplified microphone signal has a higher level than this. The deconvolution **filter** 13a gives the signal  $s(\text{sub } 1)$  an upper critical frequency of 8 kHz, thereby acting as a **band stop**, after which the signal  $s(\text{sub } 1)$  is transmitted to a first input of the...
- ...viz. the impedance converter 10b, the microphone amplifier 11b, the compressor 12b and the deconvolution **filter** 13b to a second input of the sample-and-hold circuit SH with equal **band limitation**.
- By means of a not shown monostable multivibrator MVM the signal  $s(\text{sub } 2)$  is...

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1	62	381/71.14.ccls.	USPAT	2003/06/27 13:47
2	26	381/71.14.ccls. and (bpf or bandpass or band adj3 pass)	USPAT	2003/06/27 14:03
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